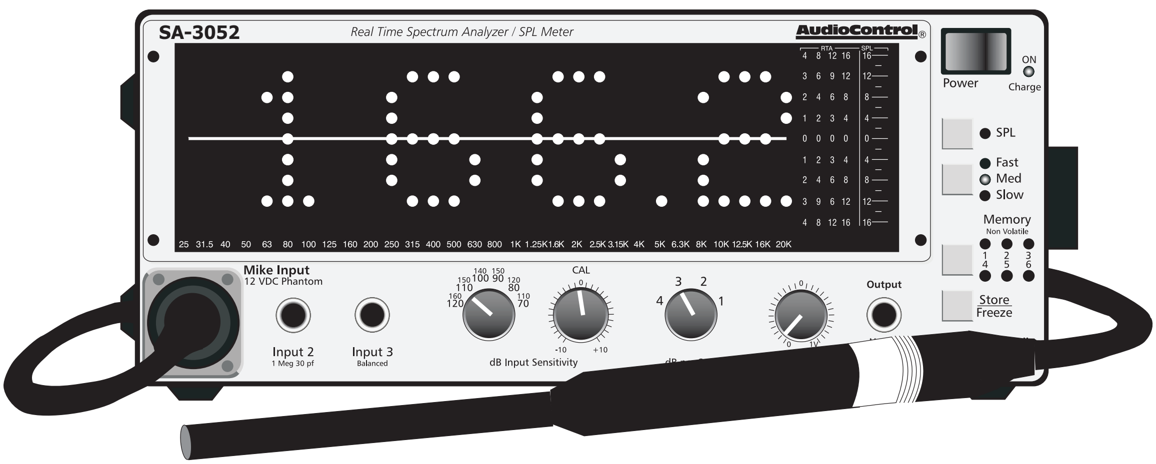
SA-3052A™
Competition Autosound Analyzer/SPL Meter

Operation Manual



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SA-3052A Owner's Manual

AudioControl®

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About This Manual

This manual describes the AudioControl SA-3052A and a few of its audio applications. The manual is divided into several major sections.

Notational Conventions

Within this manual, several different notation conventions are used to indicate various facets of the SA-3052A's features.

SMALL CAPS	Indicate a marked feature on the unit, like a control or a connector. They are also used within procedures to identify controls and switches by function.
Italics and **boldface**	Are used for emphasis. Words printed in **boldface** convey more em phasis than those printed in *italics*.

Notes, Cautions, and Warnings

Some of the text in this manual is set apart by the headings: Note, Caution, or Warning

These terms are used to denote varying degrees of awareness required by the user during installation, operation, or maintenance of the SA-3052A.

NOTE conveys information that may be helpful to the user. A note is similar to an aside during a conversation.

CAUTION indicates a potential danger to the instrument.
WARNING indicates a potential hazard to the operator.

Introduction

Congratulations on purchasing one of the world's most popular audio analyzers. The AudioControl SA-3052A is an affordable, measurement-grade, one-third octave real-time analyzer designed for audio signal analysis. Previous analyzer designs were either too costly for the average professional or too inaccurate for serious use. The SA-3052A overcomes these inadequacies by combining a state-of-the-art microprocessor-based design with modern electronic manufacturing techniques. The result is a very accurate, quick and easy to use instrument, self contained in one box.

The SA-3052A includes the following features:

- Car stereo contest scoring
- Portable battery operation
- Parallel printer port
- Completely self contained opeating system and programing
- 30 one-third octave bandwidth filters
- Fourth-order filters conform to ANSI S1.11-1986 standards
- Laboratory-grade calibrated measurement microphone
- Internal digital pink noise source
- 36 dB display window
- 9 x 30 large-format LED display matrix
- Full-screen digital SPL readout with $^{1}/_{10}$th dB resolution
- Six non-volatile memories with lithium battery backup
- Frequency response averaging for up to six stored response curves
- 20 second temporal averaging
- Peak-hold
- Balanced XLR microphone, balanced TRS ¼" phone jack and unbalanced RCA connector inputs
- 175 dB sound pressure level measurement (optional)

Test Instruments vs. Toys

There is a fundamental difference between the many real-time analyzers currently available on the market today. That difference is one of calibration. An instrument that is calibrated is capable of making measurements based on some absolute reference (absolute measurement).

An instrument that isn't calibrated can only make comparisons against a relative reference (relative measurement). For example, someone asks you, "How hot is it today?" If you step outside for a moment and reply, "Oh, a bit hotter than yesterday," that's a relative measurement. On the other hand, if you step outside and consult a certified laboratory thermometer and say "85 degrees F," you've made a measurement against an absolute reference.

The SA-3052A was calibrated at the factory to meet or exceed its published specifications. You can make a measurement with the SA-3052A and compare it with a measurement made by any other measurement-grade instrument and know that you are comparing apples with apples.

ANSI Filter Design

A real-time analyzer works by dividing the audio spectrum up into equal bandwidth parts using a set of calibrated (usually) bandpass filters. Then the output of each filter is displayed on some sort of level indicator: LED matrix, Video display, LCD, etc.

The American National Standards Institute (ANSI) oversees the establishment and maintenance of various engineering standards in conjunction with the International Standards Organization (ISO) in Europe. These two standards organizations have established standards for the performance of the bandpass filters used for acoustical measurement purposes.

For one-third octave analyzers, there are two classes of filters allowed by ANSI. These are Class II, and Class III. Class I is reserved for octave bandwidth analyzers. Class III filters are the hardest to design and manufacture and are usually only found on the most expensive (read unaffordable) analyzers. Class II filters, are easier (not easy, just easier) to design and manufacture than Class III filters. Any real-time analyzer intended for serious measurement work will use filters that at least meet Class II standards.

The filters used in the SA-3052A analyzer meet or exceed the standards specified by ANSI for a Class II, Type E filter set. Simply, they are very, very good but still affordable.

Figures 1.1 and 1.2 show two different one-third octave filters. Figure 1.1 is representative of those found in most analyzers. Figure 1.2 is an actual filter used in the SA-3052A. Note the difference in the response away from the center frequency. The inferior filter's broad response characteristic translates to potential measurement errors at frequencies removed from the band-center and undesirable interaction between adjacent frequency bands.

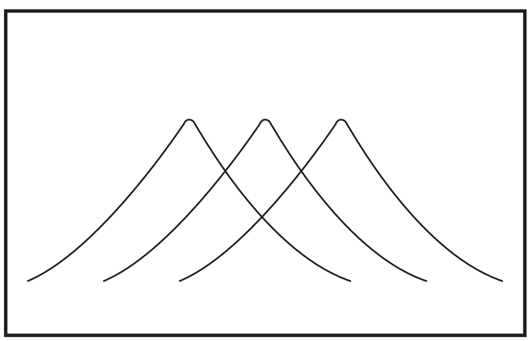


Figure 1.1. A non-measurement quality one-third octave filter

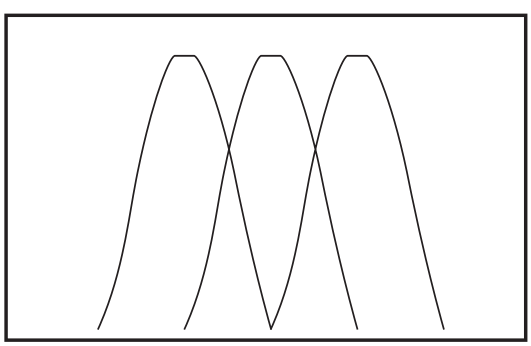


Figure 1.2. A measurement quality one-third octave filter

Applications

The SA-3052A is well-suited to many audio applications including but not limited to:

• Acoustical analysis for sound reinforcement or recording studios

• Equalizer adjustment and setup

• Crossover level setting and verification of frequency

• Gain matching and setting at various stages of the signal chain

• Movie theater system setup and alignment

• Industrial noise measurement

Special Features

Rechargeable Internal Battery Pack

The rechargeable battery option allows the SA-3052A to be used independent of AC mains power. The batteries install within the instrument and provide up to 4 hours of operation on a full charge.

The batteries are recharged by plugging the instrument into a source of AC mains power and leaving the front panel **POWER** switch set to **OFF** for a period of 9 to 10 hours. When charging, the power light glows red. When fully charged the red power light flashes occasionally.

Printer Output

The printer output option provides a Centronics parallel interface for a PC compatible printer (not provided). The analyzer-to-printer cable is a standard PC-type parallel printer cable (also not provided).

The printed output is a form with suitable blank spaces for recording the time and place of the measurement and other data, followed by the a printed representation of the real-time spectrum display on the analyzer.

On-screen score display

For quick check-ups in the shop when you don't have a printer handy, the SA-3052A will display the frequency response score on the screen. Of course if you do have a printer, the SA-3052A provides complete information about the individual response deviations on a special printout.

Automatic SPL judging

Forget about finding your stopwatch. Forget about disputes over SPL scores. The SA-3052A automatically times 30 seconds and holds the loudest SPL peak during the count-down. An LED bar ticks off the seconds for great crowd appeal.

No computer required

The SA-3052A works directly with your PC-compatible printer. Just plug your printer into the SA-3052A and you're ready to go.

SC-10 Soft Carrying Case

The optional soft carrying case protects the SA-3052A from the ravages of porta ble operation. It includes a large pocket to carry the mike and cables. There is also a hatch on the bottom of the case to allow access to the power cord connection.

Options

Customized Printout

The CP-10 option customizes the printer output form to include six lines of data for your business name, address, and other data. Since this data is ***only*** factory installed, it has the additional benefit of “branding” your analyzer in case someone wanders off with your prize analyzer.

AC-10 A- and C-weighting Filter

The AC-10 A/C weighting filter is a compact plug-in module designed to be inserted inline with the microphone. The AC-10 applies the standard A- or C-weighting function to the microphone signal.

Very High SPL option

The high SPL option allows sound pressure level measurements over 170 dB. This option includes a high SPL microphone and software upgrades.

External SPL Display

Designed for SPL contests, this option includes a large, very bright SPL display which connects to the SA-3052A. The output is in big, bold numbers, which are easily readable by the contest crowd. Two analyzers and displays may be started simultaneously by a single simulstart button (SS-10).

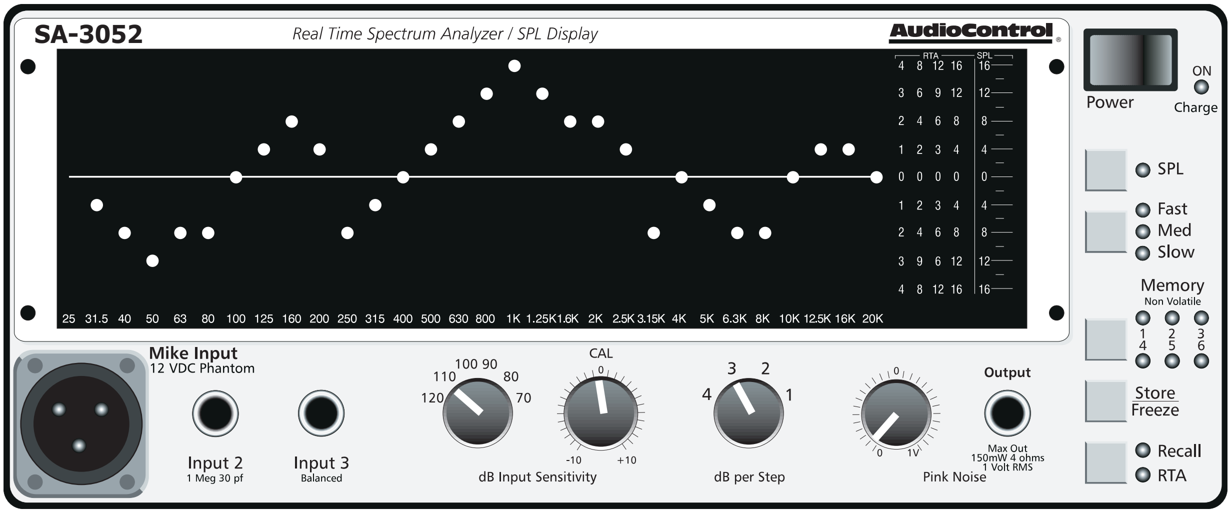


Figure 2.1. SA-3052A Front Panel

Front Panel

LED Display

The left portion of the display shows the one-third octave energy content of the input signal from 25 Hz to 20kHz. Each column within the display represents a single one-third octave band. The band centers are marked on the bottom of the screen.

The right-hand end of the display indicates the setting of the **dB** switch, and consequently the amount of change in the input signal that each LED on the left- hand side of the display represents.

The far right portion of the display indicates the sound pressure level (SPL), relative to the settings of the **dB** switch and control.

Power Switch

Depressing the **Power** switch to the right turns the SA-3052A on. Pressing it to the left turns it off.

Low Batt/Power/Charge LED

The **Low Batt/Power** indicator illuminates solid green during normal operation of the SA-3052A. When operating the SA-3052A on the optional internal rechargeable batteries, a flashing green LED indicates that the batteries need recharging. Once the Low Batt LED has begun flashing, there is approximately ½ hour of battery life remaining.

The SA-3052A has a sense circuit that prevents operation of the instrument below critical battery cell voltage. This prevents total battery discharge, which shortens the cell life.

To continue operation, the SA-3052A may be operated from the AC power line. The batteries may be recharged by plugging the SA-3052A into a suitable source of AC power, with the front panel **Power** switch set to the **Off** position for a period of 9 to 10 hours. The power LED will glow red while the batteries are charging. The minimum recharge time from the voltage-cutout state is around 2 hours, which will then operate the instrument for around one hour. Completely recharging the batteries from a discharged state (instrument dead) requires 9 to 10 hours (2 times the discharge time).

SPL

Momentarily pressing the **SPL** push-button switch toggles the **SPL** bargraph at the right side of the display window on and off. Pressing and holding the **SPL** push-button activates the full-screen digital **SPL** display. Momentarily pressing the **SPL** switch again restores normal analyzer operation. The associated LED indicator indicates the status of the **SPL** switch. The SPL bargraph display scaling is always 4 dB/step, regardless of the setting of the dB switch. The **SPL** switch works regardless of the setting of the **Recall/Rta** switch. This means that you can switch the **SPL** bargraph on and off even with a stored display.

Display Speed

The display speed push-button switch is located underneath the SPL switch and to the left of the **Fast, Med** and **Slow** LEDs.

The lighted LEDs to the right of the switch indicate the response time of the display. Press the button to change the decay time of the display. The Fast setting is optimized for looking at transients in conjunction with peak hold, the Med setting is useful for program monitoring, and the Slow setting averages 20 samples over a ½ second period. This works well for measurements using pink noise. A fourth display speed is a 20 second time average. It is indicated by the **Slow** LED flashing. During the first 20 seconds as the average is accumulating, the **Slow** LED will flash 2 times per second. Once the accumulator is full, the flashing will slow down to 1 flash per second.

Memory

The SA-3052A can store up to six different frequency response curves (including the SPL bar graph display) in its internal non-volatile memory. These memories are stored at the highest resolution of the SA-3052A, so you can scale the display with the **dB/Step** switch during memory recall. Any combination of the memories can also be averaged. Look in the Operation section of this manual for further information.

An internal back-up battery maintains the contents of the memory for periods of up to one year, even with the AC power supply disconnected. This feature allows the unit to “remember” a standard curve that may be recalled each time that the SA-3052A is used.

Pink Noise

The internal pink-noise generator is an accurate, digital, laboratory-grade test source. The pink noise output is accurate throughout the measurement range to within 0.25dB.

The maximum output provided at the ¼” tip-sleeve phone jack is 1 volt into a 600 load (unbalanced, ring and sleeve grounded), or 150 mW into a 4 ohm load.

The signal level at this connector is controlled by the knob to its immediate left.

The pink noise generator has sufficient output to drive virtually any speaker or crossover directly.

dB Per Step

This switch sets the resolution of the spectrum analyzer portion of the display. The setting represents the value of each LED in the display. Thus, a setting of 1dB per step causes each LED in the display to represent a 1dB change. At this setting, the overall range of the display is 9dB. At the 4-dB-per-step setting, each LED in the display represents a change of 4dB, with a 36dB overall display range.

When using the SA-3052A for sound system equalization, start with the 4 dB/step setting and progressively decrease the setting of the **dB Per Step** switch as the equalization process progresses. For program monitoring, the 4dB position works well because it displays the widest dynamic range.

Memorized response curves are always stored at 1-dB-per-step-resolution, regardless of the setting of the front panel switch. You can select whatever resolution you wish for a stored display when it is displayed and know that it is accurate. The same holds true for displays dumped to the printer output.

dB Input Sensitivity

The **dB Input Sensitivity** control and switch select the reference level of the curve shown in the display window. The control to the right of the input selector switch is a fine adjustment and alters the range selected by the switch over a range of ±10 dB. The normal setting for this control is the detented (click-stopped) center position.

The reference level of the display is shown on the display window by the light gray horizontal line. The actual reference level of the reference line corresponds to the dB level setting of the **dB/SPL** switch.

For example, the **dB/SPL** switch (coarse adjustment) is set to 90dB. The light gray line on the display window represents a sound pressure level of 90dB if the 10dB control (fine adjustment) is set at the center-detented position.

Input 3

A ¼” tip-ring-sleeve phone jack is used for connecting to balanced and unbalanced sources. Connect unbalanced sources by using a tip-sleeve (2 conductor) plug inserted into this jack.

This input has an impedance of 10 kohms and is suitable for signal levels from -56dBu to +36dBu. An input signal of 0dBu represents 100dB SPL.

Input 2

This is a standard audio (RCA) connector with an input impedance of 1 Mohm in parallel with 30 pF. The particular combination of resistance and capacitance allows connecting almost any audio source to this input.

This input is suitable for signal levels from -56dBu to +36dBu. An input signal of 0dBu represents 100dBSPL.

Input 1

This is a standard +12 volt, phantom-powered, balanced microphone input. The phantom power supply is intended for the AudioControl CM-10 or High SPL measurement microphones.

Ordinary dynamic microphones may also be connected directly to this input providing that they have a balanced output. Unbalanced microphones may not be used. This input is suitable for acoustical levels at the microphone ranging from 44dB SPL to 136dB SPL. The SPL display is only accurate if the AudioControl CM-10 microphone is used, or if the external microphone matches the sensitivity of the CM-10 microphone.

CAUTION: A microphone with an unbalanced output may be damaged if connected to this input. AudioControl assumes no responsibility for microphones damaged in this manner. Connect microphones with ¼” phone plug connectors to the Balanced Input 3 connector.

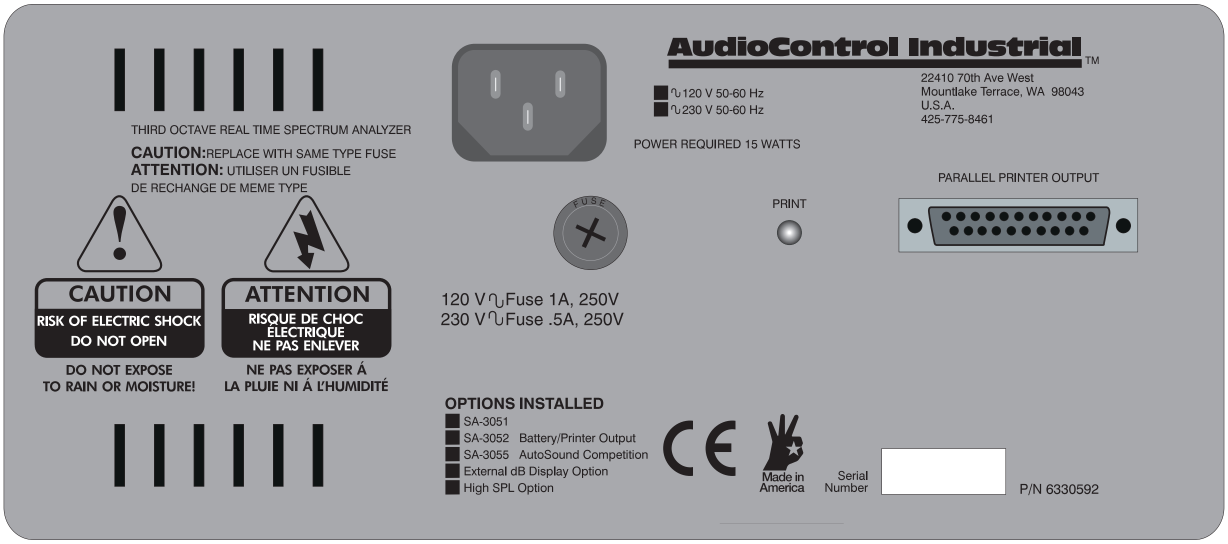


Figure 2.2. SA-3052A Rear Panel

Rear Panel

AC Mains power input receptacle

An IEC 3-prong receptacle provides the connection for AC power input to the SA-3052A.

The SA-3052A is factory supplied to operate at one of the following AC power supply voltages. The supplied configuration is marked on the rear panel of the unit. Consult the AudioControl factory for power supply conversion information.

■ 120 VAC, 60 Hz

■ 220-240 VAC, 50/60 Hz

Power Fuse

The correct fuse is listed next to the power input receptacle.

Printer Interface (printing function no longer supported)

This is a female 25-pin connector (DB-25F) used for connection to an external PC-type printer (not supplied). The connector is wired for a parallel interface printer. A description of the pin connections can be found on page 3-5.

Print (printing function no longer supported)

This momentary-contact switch is used to print the contents of the display onto an external printer.

Using the SA-3052A is simple and straightforward. This section describes the basic procedures for using the SA-3052A.

Quick-Start Information

1. Connect the SA-3052A to a suitable source of AC power. If your instrument is equipped with the battery option, the batteries must be charged before use. The correct AC power configuration is marked on the rear panel of the instrument.

2. Connect the input signal source to the appropriate input on the front panel of the instrument.

3. If you are using the internal pink noise source, connect the **PINK NOISE** output of the analyzer to the amplifier or sound system line input. The **PINK NOISE** output has sufficient power capacity to directly drive a loudspeaker to modest levels.

4. Set the **dB INPUT** switch so that signal peaks reach the upper third of the display. Adjust the **dB INPUT** control as needed. If the variable ±10 dB control is centered, then the gray line dividing the display represents the SPL setting of the **dB SWITCH**.

5. Select the display resolution using the **dB PER STEP** switch. The switch setting represents the value of each LED in the display. A scale is printed on the right-hand side of the display window.

6. Select the decay time of the display by pressing the push-button switch adjacent to the **FAST**, **MED**, or **SLOW** LEDs. The Fast setting is suitable for viewing transients. The Med setting is suited to signal monitoring and the Slow setting works well for pink noise measurements.

7. You can determine the sound pressure level at the microphone by pressing and holding the **SPL** push-button for one second until the display changes to the digital mode. This will display the sound pressure level in $^{1}/_{10}$th dB (centibel) resolution. The SPL bar-graph is always 4dB per step, regardless of the setting of the **dB PER STEP** switch.

Using The SA-3052A's Functions

Storing a Response Curve

1. Press the **STORE/FREEZE** button. The display is now frozen and ready to be stored.

2. Pressing the **MEMORY** button again steps through the six memories in sequence. The display shows the contents of each memory as it is selected.

3. The **RECALL** and **RTA** LEDs are not illuminated. This indicates that the SA-3052A is ready to store the display.

4. Pressing the **STORE/FREEZE** button again stores the contents of the display into the selected memory. The associated LED indicators display which memory is currently active.

5. The yellow **RECALL** LED illuminates to indicate that you are now in recall mode. You can recall any other memory for comparison purposes by selecting that memory by repeatedly pressing the Memory button.

6. Press the **RECALL/RTA** button to return to real-time mode.

Recalling a Response Curve

1. Press the **Recall/Rta** button until the **Recall** LED is illuminated.

2. Press the **Memory** push-button to successively view the contents of the six memories.

3. Pressing the **Recall/Rta** button again returns the SA-3052A to the RTA mode.

Averaging Several Readings

The averaging mode allows up to 256 previously stored readings to be averaged together. The averaged result is stored in memory number six, replacing its contents.

1. Acquire and store up to six response curves.

2. Press and hold the **Store/Freeze** button for one second. The yellow **Memory 6 LED** is now flashing, indicating the initiation of the averaging mode.

3. Select the first memory for averaging using the **Memory** button. Touch the **Store/Freeze** button once.

4. Select the next memory for averaging using the **Memory** button. Touch the **Store/Freeze** button once. Repeat this step for each additional memory to be averaged. You can weight a specific memory by selecting it more than once.

5. After selecting the memory locations to be averaged, press and hold the **Store/Freeze** button for one second. The **Memory 6 LED** will come on solid and the average will be displayed on the screen and written into memory location 6.

6. The contents of memory location 6 are lost when the average is calculated. The result of the averaging operation is overwritten into memory location 6.

Peak-Hold

The Peak-Hold mode allows you to accumulate the highest peak level in all of the SA-3052A's thirty display bands. This display may be stored in any memory location, averaged, etc.

1. Press and hold the **Rta/Recall** button for one second. The red **RTA LED** is now flashing, indicating the peak-hold mode is active. The display now holds the overall peak level for each of the thirty bands.

2. Pressing the **Rta/Recall** button again exits the peak-hold mode.

Peak-Hold Digital SPL

The digital SPL display may also be operated in peak-hold mode.

1. Press and hold the **SPL** button until the instrument enters digital SPL mode.

2. Press and hold the **Recall/Rta** button until the **SPL LED** starts flashing.

3. The SA-3052A is now in digital SPL peak-hold mode.

Clearing the Internal Memories

1. Press and hold the **Store/Freeze** button while turning on the SA-3052A.

Printer Operation

A standard printer interface allows making hard-copy printouts of the SA-3052A's display. The printer output is a form with space provided for logging the time and date of the job, as well as any other data. A sample is shown in figure 3.1. The SA-3052A prints special output formats when in IASCA or USAC scoring modes.

Printers Supported

The SA-3052A printer interface is designed to connect to any Centronics-type parallel interface compatible printer (dot matrix, ink jet, laser, etc.). No special printer control codes or emulations are required.

> **Note**
>
> The parallel PC printer interface standard does not require any baud rate, parity or stop bit settings. You should not have to make any special settings in your parallel printer to use the SA-3052A.

> ⚠ **CAUTION:** Do not connect the SA-3052A to a serial interface printer (Apple, Point of Sale, etc.). It will not print, and damage may be done to the analyzer and/or printer.

Printing the display is easy.

1. Connect a parallel printer to the Printer Port on the rear of the SA-3052A.
2. Recall the response curve memory that you wish to print...or...freeze the display.
3. Make sure that the printer is loaded with paper, connected to a power source, turned on, and on-line.
4. Press the rear-panel PRINT button.

```
                    PERFORMANCE MATCH
                     SYSTEM ANALYSIS
                          By:


=================================================================

Client  _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _      Date _ _ _ _ _ _ _ _
Address _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _      Telephone _ _ _ _ _
City  _ _ _ _ _ _ _ _ _ _ _  State_ _ _ _      ZipCode _ _ _ _ _ _
Description of Vehicle  _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _

System Components:
   Head Unit  _ _ _ _ _ _ _  Speakers: Tweeters _ _ _ _ _ _ _ _
   Performance Match                   Midranges  _ _ _ _ _ _ _
   Units  _ _ _ _ _ _ _ _ _            Bass _ _ _ _ _ _ _ _ _ _
   Crossover  _ _ _ _ _ _ _            Subwoofer_ _ _ _ _ _ _ _
   Amplifiers _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _


Suggested System Improvements _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _

  _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _

Performance Match Technician  _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _

=================================================================

             AUDIOCONTROL SA-3055 SPECTRUM ANALYSIS
          System Frequency Response--One Third Octave
                                                                  dB
--------------------------------------------------------------  +12  92
|                                                            |   +9  89
|              * *                                           |   +6  86
|*       * * *                 *     * *     * *             |   +3  83
|--*-*-*-----------*-------*-*---*---------*-----*-*-*-*-----|    0  80
|                    *   *         *     *               * * |   -3  77
|                      *                                     |   -6  74
|                                                            |   -9  71
--------------------------------------------------------------  -12  68
25   50   100   200   400   800   1.6k  3.15k  6.3k  12.5k 20k
     Bass                Midrange                Treble

    SA-3055 V.5.54 (C) 1977     AudioControl  (425) 775-8461
```

Figure 3.1.
Printout Example.

> **Note**
>
> An abbreviated printout showing only the RTA curve may also be printed by double-clicking (pressing twice quickly) the Print button.

Connector Pin Designations

The printer interface connector uses a DB25F connector. The connector wiring is the same as that used by PC-type computers so you can use any standard PC printer cable to connect the SA-3052A to your printer.

The connector pin designations are shown in Figure 3.2.

PRINTER		SA-3055
1	Strobe	1
2	Data Bit 0	2
3	Data Bit 1	3
4	Data Bit 2	4
5	Data Bit 3	5
6	Data Bit 4	6
7	Data Bit 5	7
8	Data Bit 6	8
9	Data Bit 7	9
11	BUSY	11
19-30	Ground	18-25

Figure 3.2. Parallel Interface Signals and Pin Assignments.

Battery Operation

A standard battery package allows the SA-3052A to be operated without connection to the AC mains supply. The battery supply has sufficient capacity to operate the SA-3052A continuously for 3 to 4 hours. The battery package uses a sealed gel-cel, lead-acid battery pack.

An integral battery charger operates when the SA-3052A is plugged into the AC mains supply. The instrument must be turned off to charge the battery.

Charging the Internal Battery Pack

1. Plug the SA-3052A into a suitable source of AC power.

2. Set the front panel **POWER** switch to **OFF**.

3. Leave the SA-3052A for a period of 8 to 10 hours (twice the discharge time).

4. The battery charger automatically adjusts the rate of charge as the battery reaches full charge. The **POWER/CHARGE** LED will glow red then flash slowly when a full charge is reached. Once this occurs, the battery charger switches to trickle-charge mode to maintain the battery at full charge. Operating the SA-3052A on AC power will eventually discharge the battery.

Getting the Most From Your Battery Pack.

The SA-3052A uses a sealed, lead-acid storage battery. These batteries are not of the Nickel-Cadmium type commonly used for rechargeable applications. Lead-acid batteries offer some significant advantages over NiCad batteries.

- Given proper charging and discharge depth, the batteries should deliver 1000 to 2000 charge/discharge cycles (4-5 years of normal operation).
- Lead-Acid cells do not have memory effects from partial discharging or charging.
- Higher energy density. The lead-acid design delivers more energy per unit volume than NiCad battery systems.
- Lead-acid batteries can be charged in less time than NiCads.
- Wider temperature range: -65 degrees Celsius to +65 degrees Celsius.

Getting maximum performance from your battery pack is simple. Follow these simple suggestions:

- Lead-acid batteries do not operate well when deeply discharged. Operate the SA- 3055 until the green LOW-BATTERY indicator begins flashing. When the indicator begins flashing, you have approximately 30 minutes of operation left. Cease operation and recharge the battery or switch over to AC power.
- A low-voltage cutout prevents operation of the SA-3052A once the battery voltage reaches this point. The low-voltage cutout could make the instrument appear to be totally dead. If the SA-3052A appears dead, plug the instrument into a source of AC power. Ensure that the POWER switch is set to the **OFF** position. Leave the instrument plugged in for at least two hours and look for the red battery charging LED to glow continuously.

If this "revives" the instrument, simply continue charging the batteries for another six to eight hours. If not, consult the factory for additional information.

NOTE:

The low-voltage cutout could make the instrument appear to be totally dead. If the SA-3052A appears dead, plug it into a source of AC power. Ensure that the analyzer is turned off. Leave it plugged in for at least two hours to charge. If the **CHARGE LED** by the power switch does not light, then you may not have AC power and the battery cells are not charging.

If this "revives" the SA-3052A, you can resume normal operation using AC mains power. Be sure to charge the batteries for another six to eight hours before using the portable battery mode.

Plugging the SA-3052A into AC mains while using the battery power may cause the microprocessor to "hang". If this occurs, turn the power off, then back on again.

CAUTION: Always store the SA-3052A with the batteries fully charged. Leaving the analyzer stored with a partial charge for extended periods (more than two weeks) can damage the battery cells irreparably. The battery charging circuitry cannot overcharge the cells, so it is best to always leave the analyzer plugged into AC power for charging whenever it is not in use.

A and C Weighted Measurements (optional)

The AC-10 in-line weighting filter provides A- or C- weighting to the microphone input of the SA-3052A. Without the AC-10, the SA-3052A's measurements are unweighted. The AC-10 uses active filters to create the two weighting curves. Power for the filter is supplied by the phantom power supply on the microphone input of the SA-3052A.

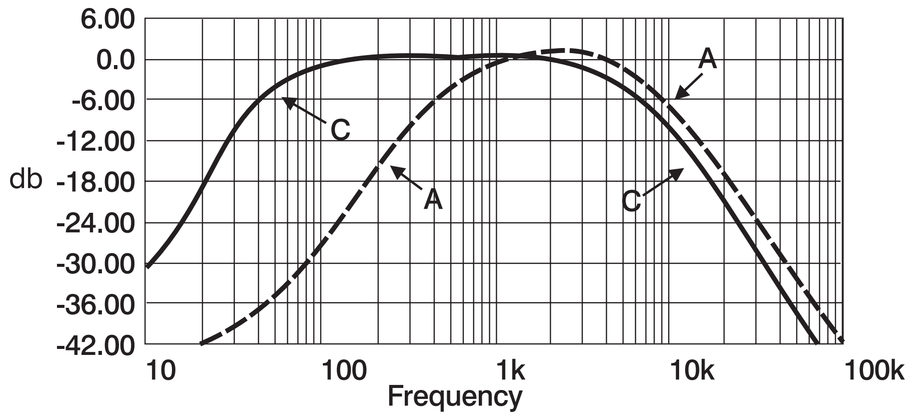


Figure 3.3. A and C weighting filter response curves

To install the AC-10:

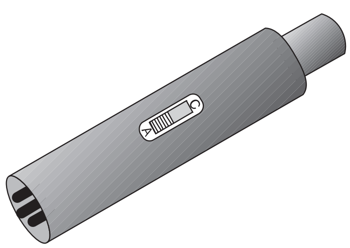
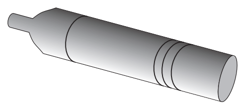


figure 3.4. AC-10 weighting filter

1. Unplug the CM-10 microphone from the mike cable.
2. Plug the AC-10 onto the end of the mike cable.
3. Plug the CM-10 into the AC-10.
4. Select the appropriate weighting curve with the **A/C SLIDE SWITCH** on the AC-10.

Getting extreme at 175dB SPL

WARNING: Exposure to sound pressure levels in excess of 90 decibels causes hearing damage. AudioControl assumes no liability or responsibility for hearing loss incurred directly or indirectly by the use of the SA-3052A. Please remember to practice safe sound.



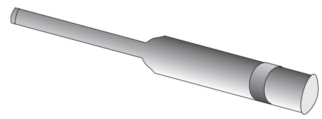


Figure 3.5. *CM-10 microphone* *High SPL microphone (no longer available*

The standard SA-3052A measures sound pressure levels up to 136dB. The High SPL option (no longer available) includes a high SPL microphone to allow measurements from 100 to 175dB. The high SPL mode only works in the digital SPL display mode. Don't use the High SPL microphone for RTA measurements. Selecting the SPL operating mode is very simple.

1. Press and hold the **SPL** button to enter the digital SPL mode.

2. The display will alternate HI and LO. Release the **SPL** button when the appropriate mode is displayed.

3. When the digital SPL mode is in the HI range, a single LED will stay lit at the left edge of the display.

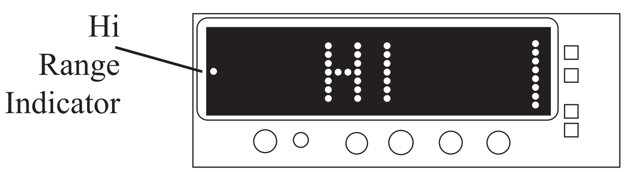


Figure 3.6
SA-3052A display showing HI SPL mode

Input Sensitivity Scale Conversion

Lo Range	Hi Range
70dB	114-146dB
80dB	124-156dB
90dB	134-166dB
100dB	144-176dB
110dB	154-180dB
120dB	164-180dB

Figure 3.7
HI SPL conversion scale

Perhaps the most important use of the SA-3052A is as an acoustical measuring tool: e.g. equalizing sound systems. This section of the manual discusses the process and rationale behind using a real-time-analyzer and an equalizer as an integral part of any sound system.

Overview (or what we're trying to accomplish)

The primary reason for using the analyzer with an equalizer is to do what our ears can't do reliably: adjust a sound system to a known reference condition. The reason that we can't do this "by ear" is our rather poor long term auditory memory. In other words, our hearing process is very adaptive. Our hearing adjusts to compensate for varying conditions, sound pressure levels, and tonal balances. When presented with a change, our sense of hearing can discern that change very readily. This is our short-term auditory memory. On the other hand, our ears are not very good at remembering things over a long period of time, especially when there are unrelated sounds between the periods of recall.

Another problem is threshold shift caused by exposure to loud sounds. Without getting into a lot of detail (don't worry, we will later), the threshold of hearing (the SPL where you can sense the presence of a sound) shifts upward when the ear is exposed to loud sound. The recovery time can be measured in hours to days, depending on the intensity and duration or the exposure. For this discussion, it's just another problem, or source of error.

For these reasons, and many more, a real-time-analyzer with a calibrated microphone is one of the most popular methods of measuring the frequency response of a sound system within an acoustical space.

In a nutshell, a pink noise source excites the space, a microphone picks up the acoustical signal and converts it to an electrical signal. The real-time-analyzer (RTA) breaks the signal up into equal octave-percentage bands, and displays the signal level of each band individually. The display could be VU meters, a CRT (video monitor or oscilloscope), or a LED bargraph. The SA-3052A uses the latter method.

In the real world, very few things are perfect, and expensive sound systems are no exception. Traditionally loudspeakers are measured in an anechoic chamber (an - not having, echoic - echoes). An anechoic chamber is a large room, with the floor, walls, and ceiling lined with acoustically absorbent material. Any sound emitted within the chamber is absorbed.

Being inside an anechoic chamber is a strange experience. First, you are standing on a suspended floor made of steel cables. If you are afraid of heights, this is just a good beginning. Since the walls absorb all sound, it is eerily quiet inside. You can actually hear your blood coursing through your blood vessels.

At any rate, an anechoic chamber is hardly the same as an average living room. However perfectly the loudspeaker measured within the chamber, putting it into an average living room changes the whole ball game. First, the floor, walls, and ceiling are anything but totally absorbent. This allows the room to have resonances, standing waves, and a sound of its own. The room interacts with our perfect loudspeaker to produce an imperfect system.

Before we go any further, you should note that equalization is a tool. Correctly applied it can do wonders. Some situations are beyond help. No amount of equalization will overcome the problems caused by reflections, standing waves, and resonance.

There is more to equalizing a sound system than simply adjusting the equalizer until the analyzer display is a straight line. It's important to know which defects you can improve on, and which ones you're wasting your time on. If this weren't so, we could train monkeys to do this, and a fair number of system installers would be looking for work.

Basic Procedure

Equalizing any sound system is basically a five-part process:

1. Listen
2. Measure
3. Balance speaker components and equalize
4. Listen
5. Trim equalization settings

The following procedure describes the basics of equalizing a sound system installed in a real-world acoustical space.

Note: If the sound system is bi-amplified, or if the speakers have level controls (like midrange and/or tweeter level controls), the first thing that you should do is to set these controls for the best overall response curve. Setting crossover level controls properly before starting is very important.

1. Connect a pink noise source to the sound system. This could be the output of the SA-3052As pink noise generator, a tape or a CD with prerecorded pink noise on it.
2. Place the microphone at the listening position. It helps to position the microphone away from any reflecting surfaces. Orient the microphone as shown in Figure 4.1.
3. Turn on the sound system. Increase the pink-noise level until the SPL is at least 10 dB over the ambient noise level. If the response curve looks terrible (huge peaks and/or valleys). try moving the microphone slightly. If the peaks or valleys go away, the problem was probably the microphone position.
4. Store the resulting response curve in one of the SA-3052As memories.
5. Repeat this process (position mike, store response) several times.
6. Recall the stored curves and derive a curve that is an average of all of them. The rationale behind averaging is to help separate out response flaws that are caused by the microphone position versus those flaws that are caused by the acoustical environment and its interaction with the sound system.

7. Now adjust the equalizer to make the average response curve flat. (level out the peaks, and smooth the overall curve)
8. Recheck your work by checking the response through the microphone positions that you used to derive the average curve. Repeat the equalization process if necessary. Your overall goal in this process is smooth response, not necessarily flat response.
9. Insert any high-frequency roll-off or bass boost needed for contouring or system voicing.

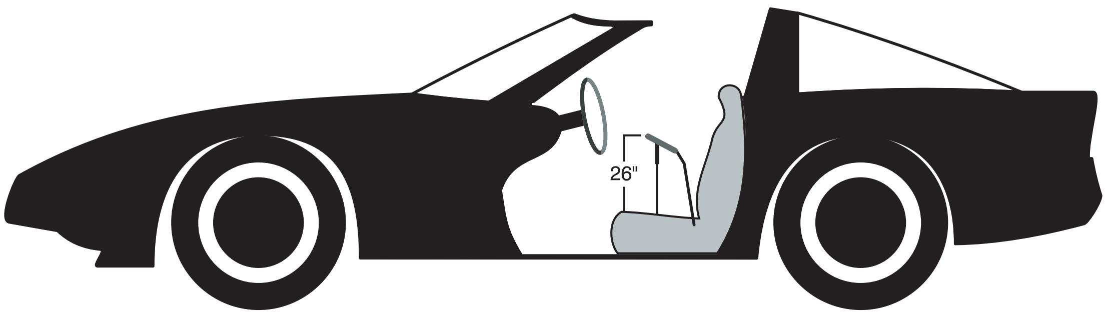


Figure 4.1 Positioning the microphone.

Now that you have an idea of the basic procedure, here are a few hints:

If the response curve really looks horrible, experiment with the microphone position. You may have just picked out a position that happened to be at a room node, or inside of a standing wave. A few inches should tell you.

The first thing that you should attack with the equalizer is the highest peak showing on the display. When you've pulled this down to the level of the surrounding terrain, find the next highest peak and pull it down. Repeat this process as necessary. Adjust the output gain control of the equalizer to make up for equalizer's loss.

If the peak straddles two equalizer controls, then split the difference between them. Unless the system and/or space are really poor, you won't need much equalization to smooth things out.

Don't try to equalize out dips into the overall response curve, unless they are quite shallow (less than 3dB). Anything more than this is a decided waste of power (3dB =2X power). Fortunately for us, our hearing is more sensitive to presence rather than absence. Thus, we'll hear the peaks in the response long before we notice the absence caused by the dips.

Once you've gotten the overall curve flattened out, listen to the system with music and speech. You'll probably want to put in some controlled high-frequency roll-off...say 1 to 3 dB per octave starting somewhere between 1 and 8 kHz.

If you're equalizing an autosound system, you'll probably need to introduce some carefully contoured bass boost to overcome the low-frequency ambient noise level in the car, and to help compensate for the Fletcher-Munson effect.

If the system is installed in a car, drive around at varying speeds to assess the effect of engine and road noise on the performance of the sound system. You may need to trim the overall equalization to suit.

Listen to the system for an extended period of time, at varying volume levels. If the overall system equalization curve isn't right, listening fatigue will let you know that you haven't finished yet. It may take several tries to get it just right.

Body Effects

When measuring, it's important to keep any reflecting surfaces away from the microphone. This includes your body if you are hand-holding the microphone. If you aren't careful, the reflection can cause response errors because of the multiple paths into the microphone. Ideally, the microphone should be suspended in free space. In reality three feet should be adequate spacing.

In an automobile, the three foot rule may turn into science fiction. On the other hand, that's the environment that the system is going to be listened to in. In this case, the rule goes out the window. You may notice more variation in the response as you move nearer to the windows. That is just another good reason for using a multiplicity of measurements points, and taking an average of all of them.

Sound Pressure Level and Hearing Loss

Although it's great to have a sound system that will achieve levels of 130+ dB SPL, it's also great to be able to hear it for a while. It's proven fact that one of the causes of hearing loss is prolonged exposure to excessive sound levels.

Research has shown that prolonged exposure to average levels under 90 dB SPL will not cause harm to our hearing. As the level exceeds 90 dB SPL, the capacity for damage increases, and the damage becomes more and more permanent. The amount of exposure that our ears can withstand without damage is related to intensity and time. We can tolerate 105 dB SPL for less time than we can continued 95 dB SPL.

The reason is the way that our ears respond to stimulus outside of the normal range of sounds in our environment. For example, if you enter a building where there is loud music playing, it may seem loud to you at first, but after 20 minutes or so, the music will seem less loud. That is because your hearing has acclimated itself to the new ambient environment. Unless you're conscious of this, you may not notice the change until you leave the building and go outside. Suddenly everything seems very quiet. You have just experienced what is known as threshold shift. After a while, your threshold will return to a level that is close (but not quite) to your original threshold. This is one mechanism of permanent hearing loss.

WARNING
Audio Control assumes no liability or responsibility for hearing loss incurred directly or indirectly by the use of the SA-3052A.

A Primer on Car Acoustics

As a listening environment, the insides of a car leave a lot to be desired. Aside from the limited space, loudspeakers need to be mounted where there is space available first and second where they can be heard. Typically this is in door panels, or in the rear deck, with the interior of the trunk used as a more or less infinite baffle.

The acoustics of a door panel can vary widely, from the rather tinny doors used on less expensive models (tinny because they tend to sound like a tin can when you slam them), to the heavy, more mechanically sound ones used on more expensive models. The quality of the door makes a difference, because after all, the door is the enclosure. In any enclosure (or listening environment for that matter), it's good practice to make the walls as rigid as possible. This helps remove cabinet resonances from the design process. In a flimsy door, the enclosure might actually disappear acoustically at the frequency where the door skin resonates (the door becomes a diaphragmatic absorber).

Next, consider the problem of the ambient noise level present in the car. Figure 5.1 shows the interior of a typical car, at freeway speeds as viewed on a real-time analyzer. Notice the rather high noise level at the low frequencies as compared to the higher frequencies. If you want to listen to music in that environment, you'll need a fair amount of low frequency boost just to overcome the ambient noise level.

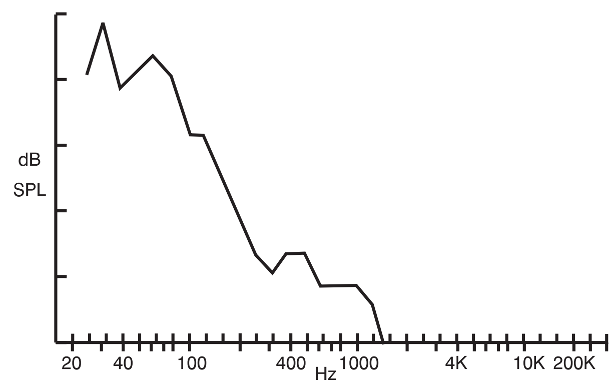


Figure 5.1. The noise spectrum inside a typical car (55 mph).

Finally, consider the interface of the installed speakers with interior of the car. Peaks in the overall response curve occur when a reflecting surface occurs at odd fractions of a wavelength from the loudspeaker or from another reflecting surface. Another peak occurs at the resonance of the passenger compartment. Of course, the speakers used are probably anything but flat and furthermore, the two sets of speakers used probably don't have identical frequency response curves (mostly due to manufacturing variations).

Statement of the Problem

Here is the problem:

• For all intents and purposes, the listener is "within" the enclosure.

• The listener's location is fairly static.

• The placement of reflecting surfaces and absorptive surfaces is haphazard from an audio standpoint.

- The ambient noise level is fairly high.
- The spectrum of the ambient noise level is not flat.
- People who sit in the front want to be able to hear, without having the people in the rear deafened, and vice versa.
- The choice of mounting locations for loudspeakers is somewhat limited, as is the amount of available space.
- The automobile was designed as a means of transportation first, and as a listening environment last.

Equalization as a Solution

Although the situation in most home or studio environments is not nearly as extreme as that within a car, all world-class studios and sophisticated home systems use carefully applied equalization. Here the equalizer provides a means of fitting the speaker into the environment, much like the final voicing that a pipe organ undergoes when it is finally installed. In recording facilities that have multiple studios, equalization provides a means for making the rooms sound alike; this provides a more consistent and predictable acoustical environment for the recording engineer.

A common anti-equalizer argument is that good speakers don't need equalization. In an ideal situation, that argument holds water, but in most cases (translate: real world), there aren't many loudspeaker/listening environment combinations that couldn't benefit from some judicious equalization.

For the home listener, there are a few options that should be exercised before using any sort of equalization:

- Change the placement of the speakers within the room to vary the bass response. You can increase the bass response by setting the speaker directly on the floor, or into a corner. You can decrease the bass response by moving the speaker away from a wall or corner, or by raising it off the floor with a speaker stand.
- The treble response can be altered to some degree by changing the amount of high-frequency absorption within the room, or by changing the settings of the speaker's mid- or high-frequency balance controls.
- The human voice is a good test signal to use to evaluate the naturalness of your system's reproduction. The spoken word can be more revealing than singing, but you should use both for your evaluation.

In a car, the positions of the speakers are limited, and most installers just can't afford to try more than a couple of different positions within the car. If you're the installer, how many times can you afford to replace the door panels or the rear deck panel if the sound isn't quite right? You could add some high-frequency absorption (say...some extra cushions), but they would probably end up in the trunk the first time that the car was used to carry passengers. Last, how many car speakers have high-frequency balance controls?

In most home stereo systems, the supplied bass and treble controls are sufficient for most user-preference adjustments. This due, in part, to the predictability of an average speaker system in an average living room. In many auto systems, the bass and treble controls provided on the head unit aren't even sufficient to overcome the shortcomings of the installation!

Some auto-sound installers have taken to tinkering with the crossover network by adjusting the actual crossover frequencies in an attempt to even out the overall response curve. While there is some merit in this technique, the primary purpose of the crossover network is to keep the individual drivers operating within their optimum frequency ranges. Thus, crossover frequency selection should be a consideration of the loudspeaker system designer and should be made **independently** of the acoustical environment in which the speaker is used.

A user-accessible equalizer is a good solution towards overcoming some of the shortcomings of the automotive environment. Within limits, it's an effective one.

- The amount of equalization needed by most installations almost negates the equalizer's usefulness, once you've adjusted it to get the overall frequency response to something resembling high-fidelity.
- Usually there is no way to equalize the two channels independently.
- There isn't an easy way for the listener to return to this starting point, unless you physically mark each control setting on the equalizer.

Bi-amplified systems have a slight advantage in this discussion. Typically you can adjust the level of the high-frequency channel in relation to the low-frequency channel (just like at home). If the system crossover point is low enough, then you can fudge the crossover point to occur at the usual 200 Hz build-up that occurs in most vehicles. This can be a real bonus in just getting the system into the acoustical ballpark.

A solution

Taking a systems approach towards the overall problem of putting a high-quality sound system inside of an automobile is the first step on the road to successfully dealing with car acoustics. This means considering every aspect of the installation from the selection of the components to the way that they interface acoustically with the interior of the car and its occupants. Unless you're willing (spiritually, emotionally and financially) to perform major surgery on the innards of the car, custom equalization of the overall system, independent of the user accessible controls is the best solution.

Custom equalization of any system allows the installer to bring the overall performance of the total system to a standardized reference point, leaving the full range of the tone controls for the owner's personal taste. That is the ultimate benefit of customized equalization.

It's a pretty well known fact that putting a good audio system together isn't all that hard. Perhaps a bit less well known is the fact that putting a superb system together is quite a bit harder. In addition to really good components, it takes a great deal of attention to detail. That's why this section is named:

Getting the Last ¼ dB

Overview

Once the system is mechanically installed, most of the really hard work is over. Now the tricky part begins. The most perfect installation can be ruined by careless electrical installation or failure to match the performance aspects of differing components with each other. Even purchasing a system from one manufacturer isn't a guarantee of component-to-component compatibility.

Noise is a given in any electronic system. You don't have to tolerate extraneous noise. However knowing the cause and cure of externally generated noise is one step towards that last quarter dB. The advent of really high-performance automotive sound systems caused the old problem of noise (in its various forms) to rear its ugly head. Even if the manufacturer has done their engineering homework, it's still really easy to forget some detail during installation and make even the quietest components seem noisy.

Consider:

- A head unit with an integral power amplifier.
- An outboard equalizer, with input and output level controls.
- An outboard power amplifier, with high- and low-level inputs.

What we have here is a potential disaster. If the level adjustments between the various components aren't made correctly, the finished system could be any of the following:

- Noisy
- Distorted
- Seemingly underpowered

In this section, we'll discuss the problem of signal level matching as a means of properly interfacing two components in a sound system. Along the way, we'll also cover the interface problem listed above. At times, this discussion may seem a bit long-winded. It is, but the risk of spreading misinformation by oversimplification is quite high, do yourself a favor and stick with us; the time you spend reading this will be well spent.

What is Noise?

In any audio system, there are several potential noise sources.

- thermal noise
- induced noise
- noise caused by a ground loop

Thermal Noise

Thermal noise sounds like the hiss that you hear between stations on your FM radio. Thermal noise occurs because mankind chooses to live 300 degrees above absolute zero. Thermal noise occurs because of the random movement of electrons caused by thermal agitation. At absolute zero, thermal agitation ceases to exist (along with the thermal noise). Unfortunately, so do we.

Just because we live at temperatures other than absolute zero doesn't mean that we can't deal with thermal noise. Careful design, along with attention to detail can minimize this type of noise. Remember...the word is minimize not eliminate.

Induced Noise

Induced noise has a variety of causes. Induced noise is sneaky. Sometimes it gets into your system through the power wiring, other times, it may sneak in via two adjacent wires. Still other times, it just gets in through the air.

Although a car is powered by a battery, (and everyone knows that batteries put out pure DC) the automobile electrical system is one of the nastiest environments known to electronic equipment. The alternator, which generates power to recharge the battery while the engine is running, puts high-current pulsating DC into the battery, which shows up every where else in the electrical system. Alternator noise sounds like a whine whose pitch is proportional to the engine speed.

Although the voltage is relatively low, the alternator wiring carries fairly high currents. Passing an electrical current through a wire is a sure-fire way to generate a magnetic field. Higher currents generate higher strength fields. Now, if you put two wires in close proximity to each other, and one of them is carrying an electric current, you can generate electricity in the other by one of two methods:

1. Moving one of the wires relative to the other.
2. Changing the magnetic field by varying the current or voltage.

Since the alternator puts out alternating current method two applies here. This is the very same principle that a transformer (not the guys on Saturday morning TV) uses. Imagine what might happen if the other wire was the lead from the head unit to the power amplifier.

The last cause of induced noise is electro-magnetic-interference (EMI). This is the same method that radio transmitters use. Basically, the sound system's wiring becomes an antenna, and the amplifiers within the system become the receiver. Too bad if you don't want to listen to the interfering station. Some examples for EMI are ignition noise or perhaps the loud buzz that you hear if you drive near a television transmitter.

Ground Loops

Ground loops are insidious. They are caused by the non-zero resistance of the wire used to interconnect the equipment. Typically, ground loops are created by a piece of equipment having multiple connections into the grounding system. More than one audio technician is bald from searching for a ground loop within a sound system.

In autosound, this problem is exacerbated by the rather callous belief of automakers and certain other folks that the metal frame of a vehicle makes a good ground. While this may be true for cigarette lighters and tail lights, applying this belief to audio systems is a gilt-edged invitation to disaster.

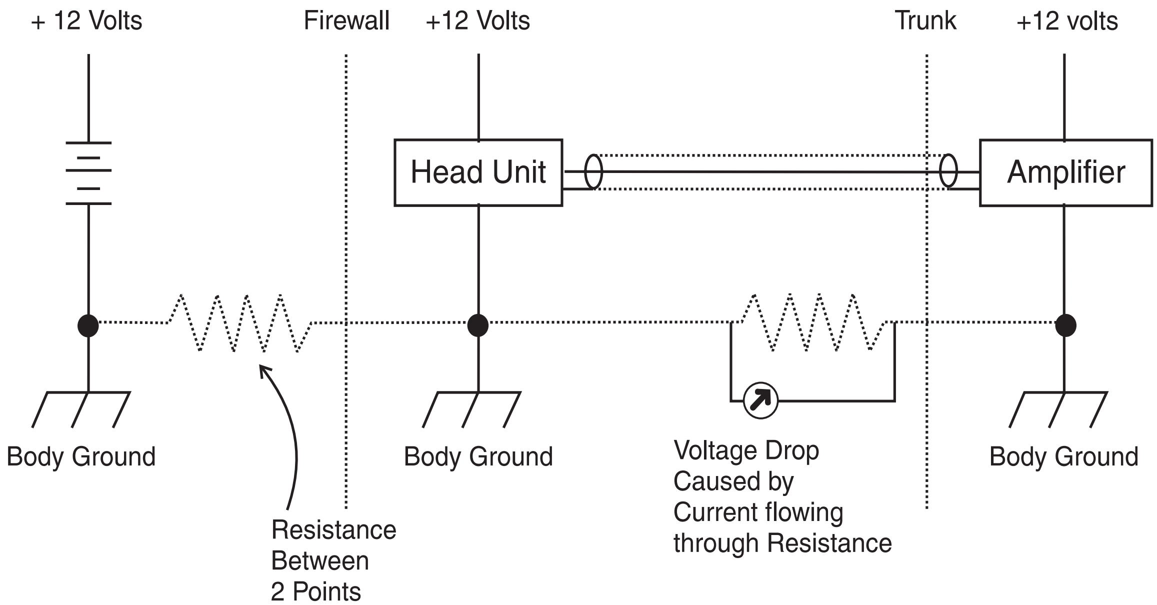


Figure 6.1 Anatomy of a ground loop.

Figure 6.1 shows what happens. Generally, the audio wiring has higher resistance than the power wiring. Since the electrical system of the car uses the body/frame structure as it's negative return, the non-zero (yes, it's small, but not small enough!) resistance of the car body allows small voltage drops to be created between various points in the car body.

The alternator in the battery charging system makes things worse because the frequency of it's AC output (alternating current...that's why it's called an alternator, not a generator) is easily within the audible range. The low impedances involved (high available current means low impedances) make filtering out alternator noise even more difficult. Anyway, these voltage drops occurring between various points in the car body usually have alternator noise riding on them, which gets into the sound system via a ground loop.

Until you've traced out a noise problem, and found it to be a cleverly concealed ground loop, the phrase "Ground isn't Ground" just sounds like random noise from another audio fanatic. All it takes is one good ground loop problem to turn the hardiest skeptic into a believer.

Chapter 6 - Getting the Last ¼ dB

The Meaning of Signal-to-Noise Ratio

Signal-to-noise ratio is a common term used on spec sheets. As common as it is, many people (both learned and not) misunderstand or misuse it. Signal-to-noise ratio is the relationship between a device's normal operating level, the noise floor, and peak clipping.

The noise floor, or the Ultimate noise

All electronic equipment operating above absolute zero generates noise. Fortunately, modern electronic design techniques can identify each noise source, all the way from the tape head in a cassette machine, to each stage in the amplifiers following it on the way to the speakers. Careful design minimizes each potential sources, resulting in optimum performance. Of course, cost is a limiting factor in the overall design process at least if most of us are to be able to purchase the device. The bottom line is the thermal noise generated by the electrical resistance of the source. In the case of a cassette machine, the source is the tape head itself. Even if the amplifiers following it were noiseless, they would still amplify the thermal noise of the head itself. This source unit noise then is the Ultimate noise.

The normal, residual noise output of any electronic device is known as its noise floor. Again it is the product of the thermal noise of the source, plus the noise contributions of each succeeding amplifier stage.

Peak clipping, or the peak ceiling

At the other end of the scale is peak clipping. Peak clipping occurs when an amplifier's output signal can no longer rise as directed by the input signal. Typically, it occurs when the peak value of the output signal attempts to exceed the value of the power supply voltages supplied to the amplifier stage.

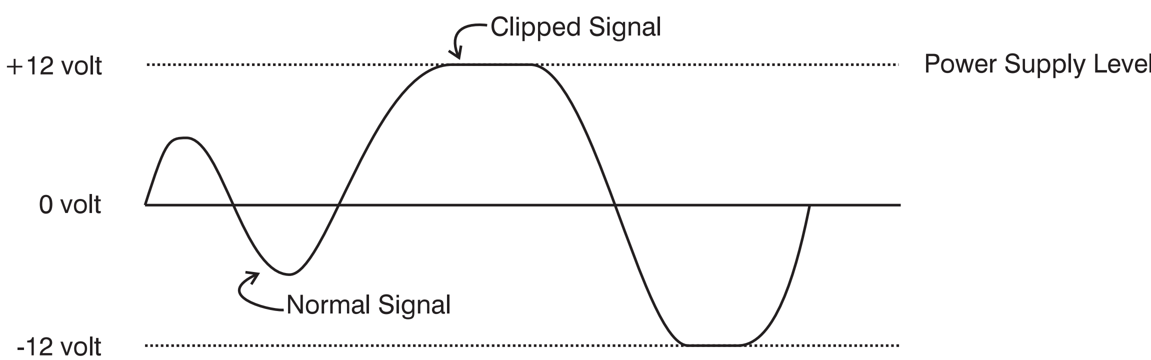


Figure 6.2 Clipping

Figure 6.2 shows this graphically. The amplifier supply voltages are + and - 12 volts. The signal can vary around ground (zero volts). up to 12 volts positive (above ground) or 12 volts negative (below ground). If the signal were to try to exceed this limit, clipping occurs as the instantaneous signal level reaches 12 volts. At this time, the output signal can go no further, and the signal peak is truncated or flattened until the signal level falls below 12 volts. The signal level at which clipping occurs is known as the peak ceiling.

Operating level, headroom, and signal-to-noise ratio

Somewhere between the noise and peak clipping is the normal operating level of a device. Looking at this relationship graphically (Figure 6.3), the distance (difference) between operating noise floor is known as the working signal-to-noise ratio. Finally, the distance between peak clipping and the noise floor is known as the dynamic range.

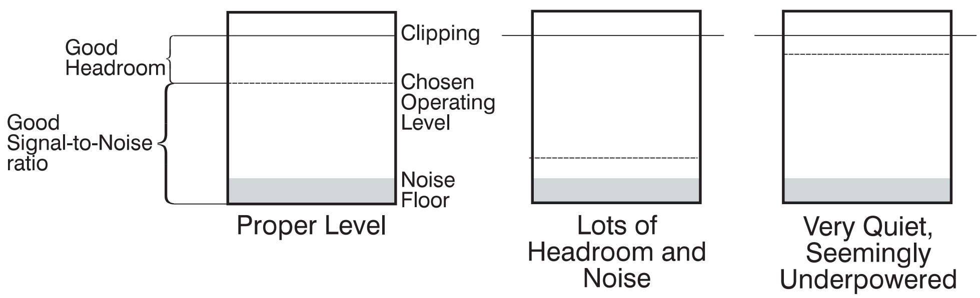


Figure 6.3 The relationship between operating level, clipping and the noise floor.

As you can see, the operating level chosen for any given device has a pretty direct effect on the working signal-to-noise ratio. Pick a level that is too high and you get premature peak clipping (but spectacular noise performance). Pick a level that is too low and you have lots of headroom, but a poor working signal-to-noise ratio. Let's put some numbers on this: a hypothetical unit, with a -70 dB noise floor, and a +10 dB clipping level.

Right off, you can see that there is a maximum dynamic range of 80 dB (+10 - -70). Now, let's explore a few different operating levels. First, we'll use a -20 dB which corresponds to a signal level of around 0.1 volt (a normal listening level on many source units). At -20 dB, the working signal-to-noise ratio is 50 dB (-70 - -20), which is okay, but not spectacular. At this same level, we have 30 dB of headroom (+10 - -20).

Now let's push the operating level up to -10 dB, about 0.3 volt. See if you can do the math yourself. The working signal-to-noise ratio is 60 dB and we have 20 dB of headroom. As you can see, there is always a trade-off between signal-to-noise and headroom. For our make-believe unit, this would be a good level to operate it at.

Some manufacturers specify the dynamic range as the signal-to-noise ratio. For the unwary person who only looks at numbers without really understanding them, this gives a truly spectacular number . Only under the most optimistic of operating conditions could the dynamic range of a device be construed as the signal-to-noise ratio. The condition under which this is true is where the headroom remaining is precisely zero dB.

The optimum operating level is a compromise between headroom and working signal-to-noise (s/n) ratio. For most systems, 60 to 80 dB of working s/n is adequate and practical. Likewise, operating level for each component in the system allows you to set the working signal-to-noise ratio/headroom trade-off.

Gain, Level, Sensitivity, and Power Relationships

Another source of error is the interrelationship between gain, signal level, sensitivity, and output power. First some definitions:

GAIN Generally this is shorthand for voltage gain. A measure of the amplification factor of an amplifier, or the output voltage divided by the input voltage. Thus, 10 volts out for 1 volt in is a gain of 10. Sometimes gain figures are converted into decibels (dB). In this case, a voltage gain converts to a voltage gain of 20 dB It's handy to use decibels for gain calculations because the only math you need is simple addition and subtraction (once you do the decibel conversion).

SIGNAL LEVEL This is the strength of a signal, measured in volts. Again, sometimes signal levels are measured in decibels, which states them as a ratio of two quantities: the signal, and a known reference level. Signal levels are also referred to by their ability to drive an input: line level, speaker level, etc. While the voltage present may be the same, the current (amperage) capability may not be the same. Basically, this is what is different between a 1 volt line level signal and a 1 volt speaker level signal.

SENSITIVITY This is a measure of the input signal requirement of a device for some stated output level. Sensitivity is intimately related to gain. If an amplifier has a gain of 10, and it's rated output level is 10 volts, then it's sensitivity is stated as 1 volt. It's important to remember that low sensitivity means a higher voltage is necessary to drive the unit, while high sensitivity means less voltage is required. Note that sensitivity has nothing to do with power output.

POWER OUTPUT Power represents energy, which can do work (like moving a loudspeaker cone). Power is always measured in watts. Power has nothing directly in common with sensitivity. It is equally possible to have an amplifier that has low sensitivity, but high power output as well as one that has high sensitivity, but low power output.

For example, let's take a system where we have a power amplifier that has relatively low sensitivity (1.5 volts for full output). We'll hook this up to a head unit that has a rated output of 0.25 volts. What is the end result? Chances are that the system won't ever realize the full potential of the power amplifier. Why? Because the head unit is giving up the ghost long before its output has reached the level required by the power amplifier for full output. What is needed here is an amplifier with higher sensitivity.

Now, let's take an amplifier that has adjustable gain. Remember that gain and input sensitivity are intimately related...high gain means high input sensitivity (lower voltage required for full output) while low gain means low input sensitivity (higher voltage required for full output). We'll hook this up to an amplified (capable of delivering 4 watts into 4 ohms) receiver. Assume that the receiver is a bit marginal in the residual noise department. If you want to get as much out of this setup as possible, then you're going to have to find the right relationship between the gains of the two components.

If you pick a high sensitivity setting on the amplifier, then the receiver's volume control will be barely open before things are too loud. Furthermore, since the receiver's electronics always generate some minimum amount of hiss, even with the volume control all the way down, the system will be hissy.

If you pick a sensitivity setting on the amplifier that is too low, then the receiver's amplifier will clip before the power amplifier does. Now the system will seem under-powered.

The right sensitivity setting on the amplifier is one where both amplifiers (the receiver's and the external power amplifier) clip simultaneously. This way, the receiver's amplifier runs at a fairly high level, which helps improve the working signal-to-noise ratio, but not at such a high level that it can't drive the external amplifier into clipping before it goes into clipping itself.

The old volume control myth

A common old wives tale is the belief that the percentage rotation of the volume control represents the same percentage of output power. Here's the way it goes: "I've got so much power that I only need to turn up the volume to 10 o'clock (20% rotation)". Since the volume control generally adjusts the overall gain of the system, there is one unique signal level where wide open (100% rotation) represents full power output from the power output from the power amplifier. Since most volume controls are followed by gain (to make interfacing varied signal sources easier), the average system reaches full power long before 100% rotation is reached. Thus, the amount of volume control rotation required to drive the system to full output depends on:

1. The signal level present at the input of the volume control.
2. The amount of gain following the volume control.
3. The sensitivity of the power amplifier (better low than high, from a noise standpoint).

Why bother? A good reason is to make sure that the signal going to the power amplifier is as strong as possible as soon as possible. This helps to subjectively reduce noise that may be induced between the head unit and the power amplifier by making the signal in the cable stronger than the induced noise.

Another reason is to increase the amount of usable range that the volume control has, which is especially useful with units that have detented (click-stepped) volume controls. The volume control taper (rate-of-change vs amount of rotation) at the low end of the control is necessarily coarse. After all, most people like to be able to shut the sound off once in a while. A system that is too loud when the control is three clicks from off isn't very useful unless the owner is a hardened volume freak. Remember to explain to the owner that even though the control must be turned "way up", that they are still getting full performance out of the amplifier.

Minimizing Noise in Autosound Systems

Now that we've established some of the causes of noise in any audio system, let's apply them to a typical autosound system. Before we start, let's make some assumptions:

1. The manufacturer did their design homework and the unit represents the best that they could do, considering their design constraints.
2. The unit meets its published specs.

Identifying noise sources

Identify the potential noise sources in the system by their characteristic sound is the first step towards curing a noise problem. All the noise filtering in the world won't help unless you can identify the source of the noise. Once the source is identified, then you can consider the cure (bypass capacitors, chokes, lead dress, or dynamite).

ALTERNATOR Noise sounds like a whine whose pitch varies with engine speed. It can be caused by poor or careless grounding, ground loops, poor wiring practices, factory wiring, old alternators or bad level matching. Bypassing the alternator output with a capacitor may help.

IGNITION Noise is a ticking sound that changes with engine speed (the clicks get closer together). It can be caused by poor or careless grounding, ground loops, poor wiring practices, or bad level matching. Resistor spark plugs may help, especially when tuners are affected.

TURN SIGNAL Noise is a clicking sound synchronized with the turn signals. Make sure that the auto body ground connections to the lamps are secure. Bypassing the input and/or output to the flasher unit with a capacitor may help. Other possibilities include poor or careless grounding, crash damage, ground loops, or bad level matching.

DASH LAMP DIMMER Noise is a buzzy sounding whine whose pitch varies with the dimmer setting. You'll probably only find this in newer cars. Try bypassing the dimmer input lead with a capacitor and/or different grounding points for the autosound system. Bypassing the dimmer output lead may cause the dimmer to overheat because the dimmer's output is effectively high-frequency AC, and the bypass capacitor tends to look like a short-circuit as far as the dimmer is concerned.

HORN Noise is a buzzing sound, synchronized with the horn (how did you guess?). Again, a bypass capacitor should fix this too.

AMPLIFIER POWER SUPPLY

Noise like a nasty buzz, that isn't affected by engine speed and is usually affected by some or all of the system controls. High powered amplifiers sometimes cause problems because the DC to DC converter inside of them radiates EMI (electromagnetic interference) into the electrical system of the car. Sometimes (especially when the units are mounted on a board in the trunk) the manufacturer will leave the chassis of the amp floating (not connected). This turns the amp chassis into an effective antenna. Connecting it to the autobody ground will turn the amp chassis back into a shield, which is what it should be. Some amplifiers have a switch to select an internal or external ground for the amp chassis.

Figure 6.4 shows how to bypass the alternator output. It's important to put the bypass capacitor as close as possible to the source of the interference. Bypassing other electrical devices is very similar. Make sure that the ground connection (the vehicle body is the place to use) to the capacitor is solid and clean. If you have an ohmmeter, measure the resistance between the capacitor and the vehicle body. It should be less than 0.3 ohm.

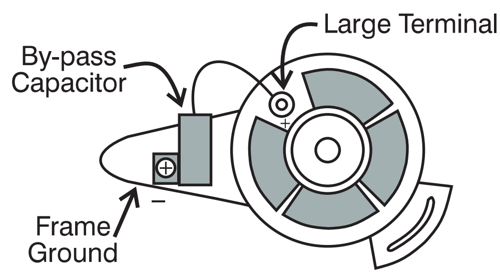


Figure 6.4. Bypassing the alternator output.

Figure 6.5 shows using a series choke and bypass capacitor to deal with more severe cases of interference. The choke raises the impedance of the power supply system at audio frequencies so that the bypass capacitor can short it out to ground. It's a good idea to use a lower capacitance mylar or paper capacitor that is shunted with a generously sized electrolytic capacitor.

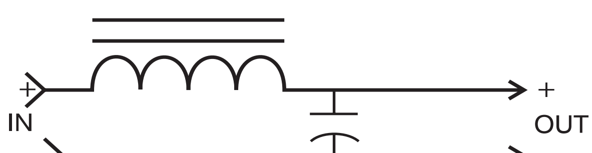


Figure 6.5. Using a series choke and bypass capacitor.

Grounding practices

CAUTION
Ground Isn't Ground!

Remember those words. Any time that you think that grounding is a trivial matter recite those three words.

The basic reason that we ground anything at all is to supply a negative return for the electric current that operates a device. In boats we use wire and have many fewer problems, if you're an automaker, basically anything that is metal and somehow associated with the vehicle body is fair game. If you're a state-of-the-art autosound installer, that belief is the first and last mile on the road to ruin.

A second reason that things are grounded is so that their metal enclosures can act as a shield for incoming EMI. Remember...the metal enclosures only act as a shield when they are grounded. This helps keep portions of the circuitry that weren't designed to be radio receivers from becoming radio receivers.

Ground loops cause problems by allowing multiple paths into the grounding system. This wouldn't be such a problem if it were not for the finite and measurable resistance of the ground system. When you combine the multiple current paths into the ground system caused by the automaker's indiscriminate definition of the word ground and the finite resistance of the ground system you get voltage drops. These voltage drops cause trouble when they get into an audio system via a ground loop. What happens is that the audio system tends to amplify any noise present in the ground system. The net result is noise.

Here's some (heh-heh) ground rules:

1. There should be one and only one path to the negative side of the vehicle electrical system. If you elect to use the vehicle body as this point, scrape the paint from under the contact point, and use an internal star lockwasher on both sides of the terminal lug. This grounding method is known as a single point ground.

   In extreme cases, you may need to run a heavy gauge wire to the battery's negative terminal. Alternatively, you could use the point where the battery negative hits the vehicle frame. Again, clean everything and use toothed lockwashers. If the connection point is subject to vibration (like on the engine), then be sure to use stranded or braided wire (braided is better), and leave enough slack so that the entire length of wire can vibrate. This will help keep it from breaking.
2. If you mount everything on a board, connect the chassis of each component to a good, solid electrical ground. The vehicle body is a good choice here because what you want is a good RF (radio frequency) ground. The wire running to the negative side of the electrical system has inductance which raises it's impedance at high frequencies. In effect, at radio frequencies, the impedance could be high enough that wire would look like an open circuit. Braided wire works best here because it has low inductance.
3. It's good insurance to use an ohmmeter to make sure that the chassis of each component isn't already tied to its internal electrical ground. Sometimes the case is tied to circuit ground through a low to moderate value resistor. If the case to

negative wire resistance is greater that 10 ohms, then it's okay to connect the unit's case to the autobody ground without causing a ground loop.

Some manufacturers put a switch on the unit to allow you to pick where or if the chassis is grounded. If you supply the connection the chassis of the unit make sure that the switch doesn't supply one too.

4. Some premium connecting cables have oversized connectors. Make sure that the shell of the connector doesn't inadvertently contact the chassis of the unit that it is plugged in to. If this happens, you've got a ground loop on your hands.
5. When you think that you've got everything right, turn everything on, and disconnect the antenna and the negative connection to the vehicle's electrical system. If everything quits, you've got a single point ground.

Remember: **Ground isn't ground**

Wiring Practices

After grounding, the rest is relatively easy, Here's a few hints:

Avoid running audio wiring along (parallel to) with high current power wiring. The improvement in noise pickup is proportional to the square of the distance between the wires. This is more important when applied to line level audio wiring rather than speaker wiring.

For high-powered systems, run heavy gauge wire directly to the battery for the power amplifiers. How heavy? That depends on the distance of the run, and the current demands of the amplifiers. You can't err by making the wire too big.

For really high-powered systems, or if the run to the amplifier location is particularly long, you may want to run separate power wiring for the power amplifiers (high current drain) and for everything else (relatively low current drain). High-powered amplifiers usually have a DC-DC inverter inside them and the inverter can radiate noise back up the power wiring. Putting the amplifier on its own wiring helps by forcing the noise to have to travel back to the battery before it can get to the sensitive low-level stuff (head unit). The battery is almost a dead short as far as the noise is concerned, so it makes an effective noise filter.

It may help to add an RF bypass capacitor at the amplifier end of the power cable to autobody ground. It may also help to add a second bypass capacitor from the positive side of the cable to the negative lead that runs back to the electrical system ground.

Bad cases of alternator noise interference will probably require series chokes in the positive power lead into the system. It may be easier (economical as well as electrically) to filter each unit in the system separately than to try to find an interference choke with enough current capacity to handle an industrial-strength power amplifier.

Don't use the auto body ground as the ground return for the speakers. They should connect to ground via the power amplifiers output connections.

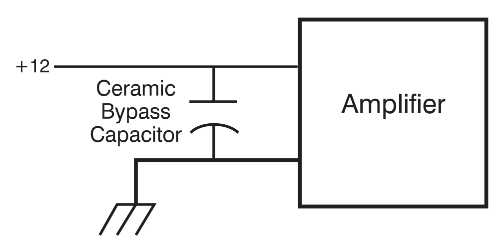


Figure 6.6. RF bypassing the power supply lines.

Setting Voltage Levels In Car Stereo Using the SA-3052A RTA

Why Level Match?

Usually, high voltage levels in audio systems, give better signal to noise ratios. You want high signal to noise levels so you do not hear the thermal hiss which is present in *all audio systems* (sometimes even worse in digital systems). As shown by Diagram 1, you need the maximum signal level to be below clipping but as far as possible above the noise floor.

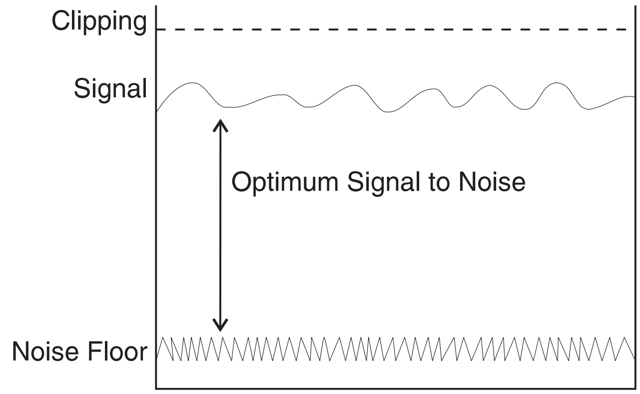


Diagram 1

Without a bunch of fancy instruments, how do you set all the levels in the sometimes numerous components in a car stereo system? The answer is as close at hand as your trusty real time spectrum analyzer.

How to Use the SA-3052A to Set Levels

1. Connect SA-3052A as shown (Diagram 2) with one channel of the output of the unit you are setting plugged into Input 2. Note: you will need to unplug the microphone or any other inputs.

2. Put the SA-3052A on digital SPL (hold the SPL button in for 2 seconds) with input sensitivity level set according to voltage goal and table on the next page.

3. Play pink noise through unit you are setting. The best way to do this is to use a pink noise CD or tape played from the source unit. You can use the pink noise built into the SA-3052A. For a pink noise CD, our recommendation is IASCA Disc No. 2.

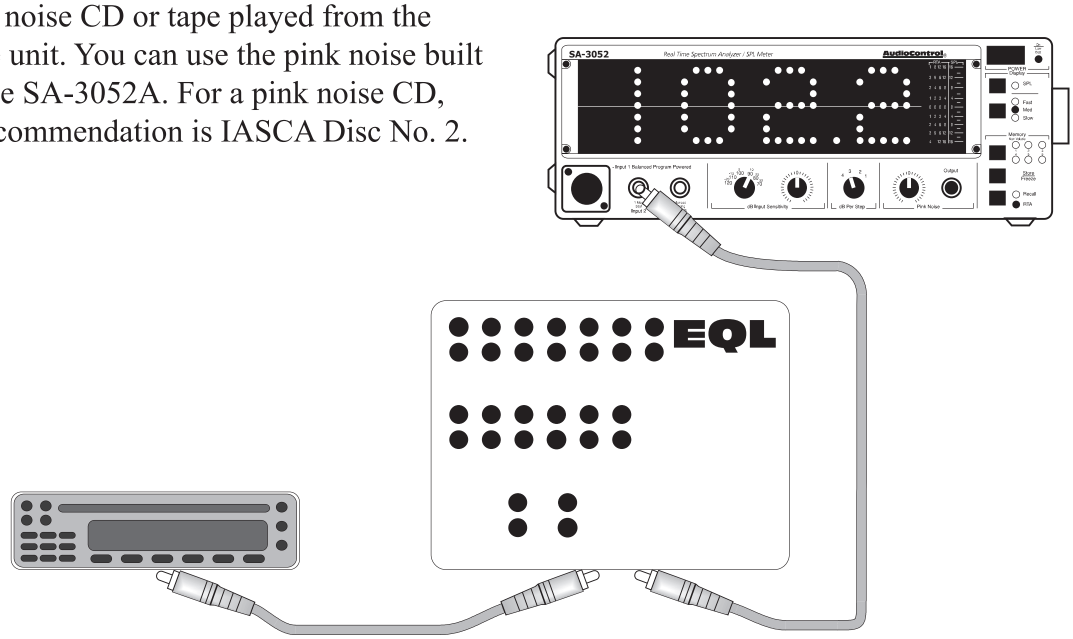


Diagram 2

4. First, set one channel then the other. All left/right pairs of channels should be the same. How high should you set the output? The answer to this is to base your settings upon the manufacturer's specifications for maximum output level. Note that decibels are logarithmic so there is a big improvement in going from ¼ volt to 4 volts and little difference between 4 and 6 volts.

5. You should start at the first unit in the signal chain and work you way toward the amplifiers. See rules of the road below for some additional advice.

Converting dB SPL on a SA-3052A into Volts

As noted before, the logarithmic nature of decibels means that the most improvement comes from raising a very low voltage to 3 to 4 volts. As you can see from this chart, an extra one-half volt up around 7 volts doesn't really count for much more signal to noise. (Amazing fact to impress your friends with; 170 dB on this type of measurement is 2,818 volts!!)

dB SPL	Volts RMS*
81	.1
91	.3
95	.5
101	1.0
105	1.5
107	2.0

dB SPL	Volts RMS*
111	3.1
113	3.9
115	4.9
117	6.1
119	7.8
121	9.8

*Full Range Using Pink Noise

Rules of the Road

1. Pay attention to maximum outputs levels from the manufacturer and don't exceed them. There is not much to be gained by pushing the envelope above 4 or 5 volts. If you run any unit in the signal chain into clipping, you will defeat the advantages of the higher signal to noise in the other components.

2. Don't drive the SA-3052A with a source impedance higher than 2k ohms. This should not be a big problem as almost all car stereo components have output impedances below 2k ohms (the AudioControl components output impedance is usually a wonderfully low 150 ohms).

3. If you want to check the output of an amp, see the procedure later in this manual.

4. If you turn up the gain and the dB reading on the SA-3052A doesn't change, you are clipping something. While clipping is usually associated with amplifiers, you can clip any unit. Particularly, be alert to a component with a lower maximum output level after one with a higher capability. (Hint: if you use AudioControl components together, you will not have this problem.)

SPL Adjustment Factor for Pink Noise after a 2-way Crossover

Crossover Frequency	High Pass	Low Pass
60	+.5	+9
90	+1	+8.5
150	+1.5	+7
270	+2	+6
1000	+3	+4
3500	+4	+2.5

Example: 150Hz crossover, SPL reading on low pass is 94. What is the voltage?
Answer: 1.0 volt (94 + 7 = 101dB adjusted)

5. Get your signal levels as high as possible as early as possible. Frequently the component with the lowest output capability is the head unit. Since the system is *only as quiet as the least quiet piece* (also known as the weakest link theory), carefully select all your components. See the tables below for more on the importance of this.

6. Turn your amplifier sensitivities down (where the volume is the lowest). Then drive a higher voltage signal into them. The result will be the same SPL/ loudness (because the wattage of the amplifier does not change) but with less noise.

7. We know you are going to get the best results using AudioControl Performance Match components as their maximum output levels are very conservatively rated at 7.5 volts.

High Signal to Noise System

Component	Mfg. Spec. Maximum	Adjust to
CD Unit	4 v	3 v
EQT	7.5 v	6 v
24XS	7.5 v	5 v
Amplifier (Input)	5 v	5 v

Crossover is weak link and limits performance of system

Component	Mfg. Spec. Maximum	Adjust to
CD Unit	4 v	3 v
EQL	7.5 v	6 v
Brand Y Crossover	2 v	1.5 v
Amplifier (Input)	5 v	1.5 v

Chapter 7 - Contest Scoring

Competition Scoring

Introduction

Welcome to computerized competition judging. If you're reading this, then you are part of the growing numbers of people who have discovered the challenge of the competition-sanctioned sound off contests. Originally, sound-offs were simple "Crank-em-Up" contests. Now the criteria for even "placing" have become more stringent than ever. Installation aesthetics, creativity, frequency response, imaging and sound pressure levels all have to be considered.

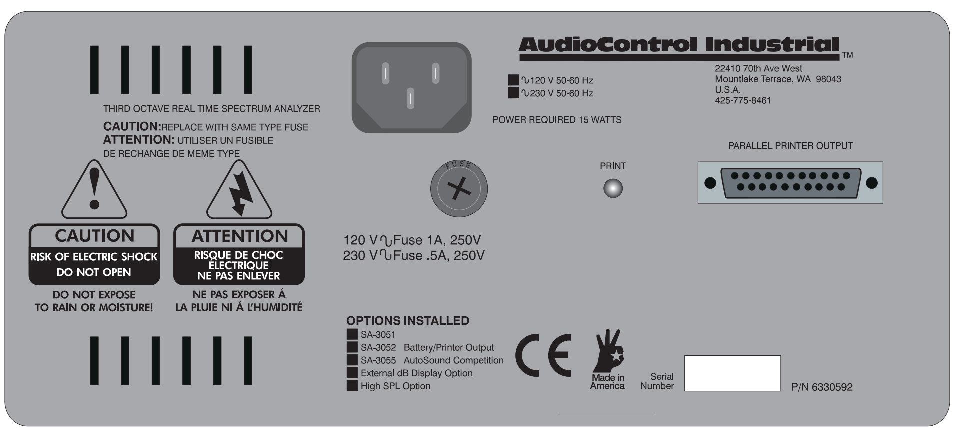
In the past, the frequency response and SPL portions of competitions were scored by hand. The amount of effort required to score even 50 cars was phenomenal. Something had to be done. Necessity (or at least, dislike of work) has always been one of humankind's greatest sources of innovation.

The AudioControl SA-3052A makes computerized scoring easy and affordable. All you need is a SA-3052A and a parallel-type printer. The SA-3052A measures the frequency response and SPL readings, calculates the score, and prints everything out. Cool!

OPERATION

RTA / SCORING mode

When you turn on the SA-3052A analyzer, the display will show the current operating mode. It will either flash '-RTA-' for normal frequency response measurement mode, 'IASCA' for competition scoring per IASCA rules or 'USAC' for USAC rules.



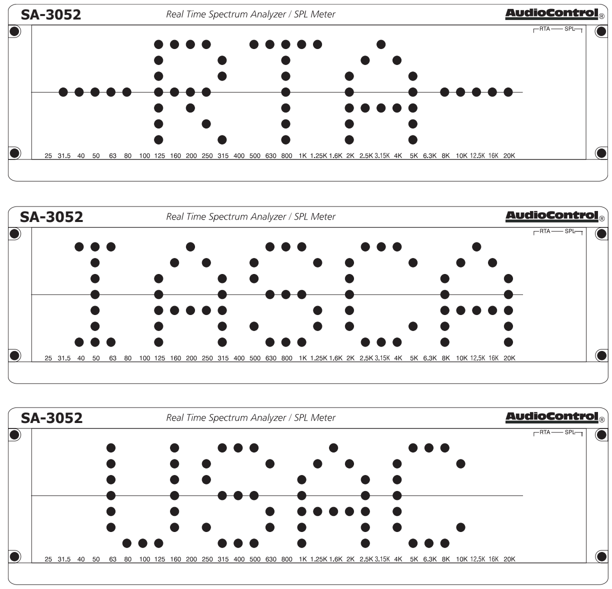
To change the mode:

1) Turn the SA-3052A off

2) Press and hold the PRINT switch on the rear of the analyzer.

3) Turn the SA-3052A back on

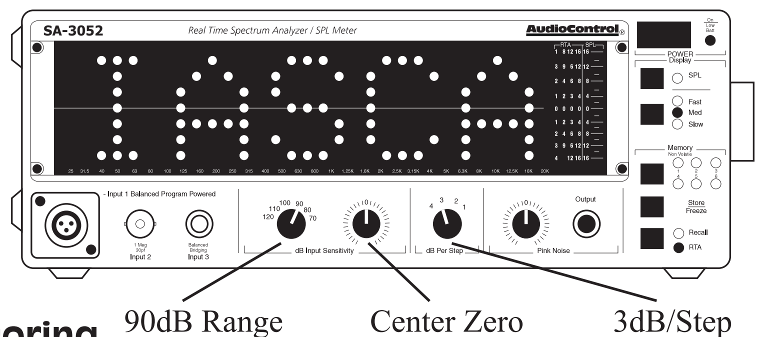
4) Release the PRINT switch when the appropriate mode is displayed. The analyzer will stay in this mode until you change it.



NOTE: The unit must be in either IASCA or USAC mode for the scoring to work.

SA-3052A control settings

For proper operation in the scoring mode, the controls on the SA-3052A must be set as shown in the figure at right. If the dB per step switch is not set on ‘3’, you will get the error message of “3dB” on the display.



Frequency response scoring

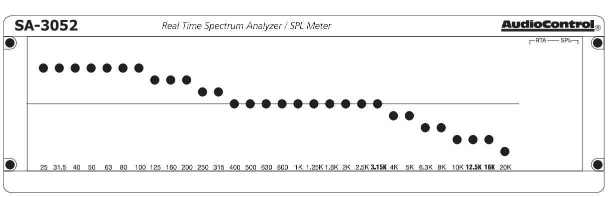
Few things are simpler than using the frequency response scoring in the SA-3052A analyzer. There are just two steps:

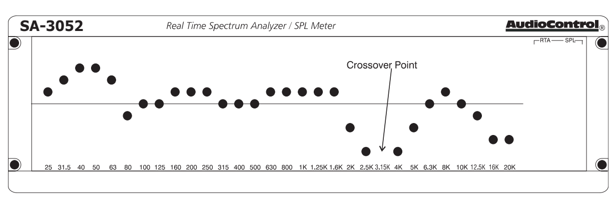
1) Press the STORE/FREEZE button or recall a memory.

2) Press the PRINT button on the rear of the SA-3052A.

Presto, a completely scored printout (assuming you have a printer connected to the SA-3052A). The SA-3052A will also display the response score.

You don’t have to wait until a competition to use the scoring feature of your analyzer. This thing is so simple to use, you should bring it out anytime there’s a customer in





the parking lot that could stand some improvement in their system. Show them where an equalizer, amp or speaker might improve the sound of their car.

Here is a recap of the RTA judging procedures

1) Place the mike (mounted on an *IASCA* specification stand) in the driver's seat of the car.

2) Make sure the engine is off and all doors, windows and other openings are closed.

3) Set the RTA controls to MED Display Speed, 3dB per step, 90dB range, and center the ±10dB fine sensitivity control.

4) Adjust the pink noise volume level of the system until the RTA is reading 90dB SPL inside the vehicle.

NOTE: If the sound level inside the car (with the stereo *off*) is greater than 80dB SPL, actions must be taken to lower the noise level in the judging area. Otherwise, RTA and sound quality judging will be affected.

5) Once the system volume is set to 90dB SPL, switch the analyzer to the RTA mode and adjust the input sensitivity control to 80dB. Make certain the display speed is on MED.

6) Adjust the fine (±10dB) sensitivity control on the RTA until the top of the displayed curve is just flickering below the top LED row.

7) Set the RTA display speed to Slow. If the curve runs off the top of the display, reduce the ±10dB control slightly.

8) Freeze and store the curve into memory.

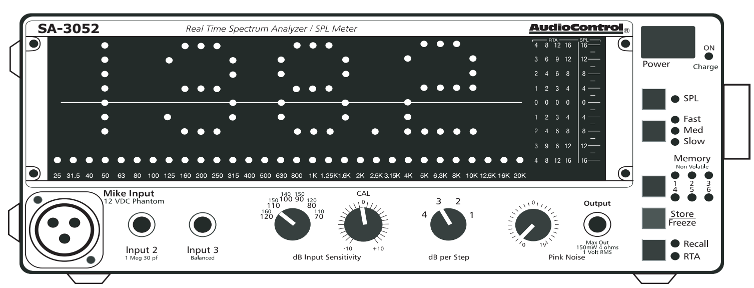
9) Print the score sheet and log the score on the safety sheet.

SPL scoring

The SA-3052A features an automatic 30 second timer and peak hold for the SPL section of your competition judging. To operate the SPL judging:

1) Press and hold the SPL button. This will switch the display to the digital SPL mode. NOTE: If you have the SPL-175 option, set the SPL range to match the microphone you are using (LO or HI).

2) Set the input range switch on the SA-3052A to 120dB. This will allow SPL measurements from 104dB to 136dB. If you suspect the car will not reach 104dB, set the input range switch to the 110dB position.



NOTE: Make certain the ±10 fine sensitivity control is set in the center '0dB' position or the display will be incorrect.

3) Press the FAST/MED/SLOW button. All three speed LED's will begin to flash. This indicates the timer is standing by.

4) When you are ready to crank the system and start the timer, press the FAST/MED/ SLOW button again. A thirty second bargraph will start counting down across the bottom of the display.

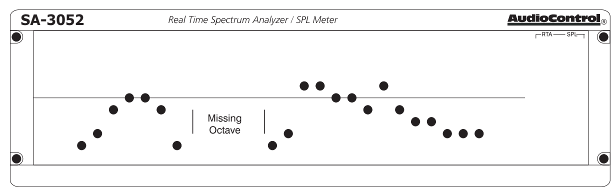
5) After 30 seconds, the three speed LED's will come on and the display will freeze. This is the maximum SPL during the countdown.

ERROR MESSAGES

If the HAL-30 determines that something isn't correct scoring a curve, an error message is displayed on the screen. Once you have fixed the problem, press the RTA/RECALL button to clear the error message.

ERROR: MISS - Missing Octaves

More than one octave is missing in the frequency curve being judged. Although you can probably infer the trend from the slope of the curve, the SA-3052A would rather leave such things to humans. They excel at reading between the lines.



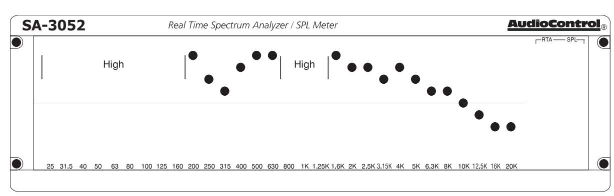
In this example, there are a couple of possible explanations for the missing octave:

1) **Low input level:** the missing bands could be lurking just off the bottom of the screen. Try adjusting the ±10dB fine input control up to see if more of the curve can be brought onto the display without any of the bands going off the top.

2) **System failure**: A "gap" in the frequency response can be the result of part of the system not playing. The mid-bass amp, crossover or speaker may have failed. Have the competitor double check their system to make certain everything is working correctly. It could be a simple blown fuse causing this hole in the response.

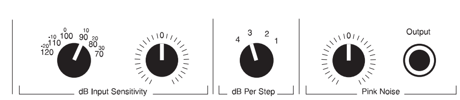
ERROR: HIGH - High reading

The HAL-30 has determined that several frequency bands are off the top of the scale. Lower the system volume to get the curve onto the display and repeat the measurement.



ERROR: 3dB - Wrong display range used

The dB per step switch on the SA-3052A *MUST* be set on 3dB/Step for the scoring option to function. Set the dB per step switch to 3 and press the print button again.



A TROUBLESHOOTING GUIDE

NOTHING PRINTS

Is the cable plugged into printer and SA-3052A?

In the heat of a competition, people can bump into the judges bench and knock a cable loose.

Does the printer have a parallel interface?

Apple or Point-of-Sale-type printers usually have a serial interface and will not work with the SA-3052A. An easy way to double check is to look at the connector on the printer. It should have 36 *flat* connections rather than the 25 round connections like the plug on the rear of the SA-3052A.

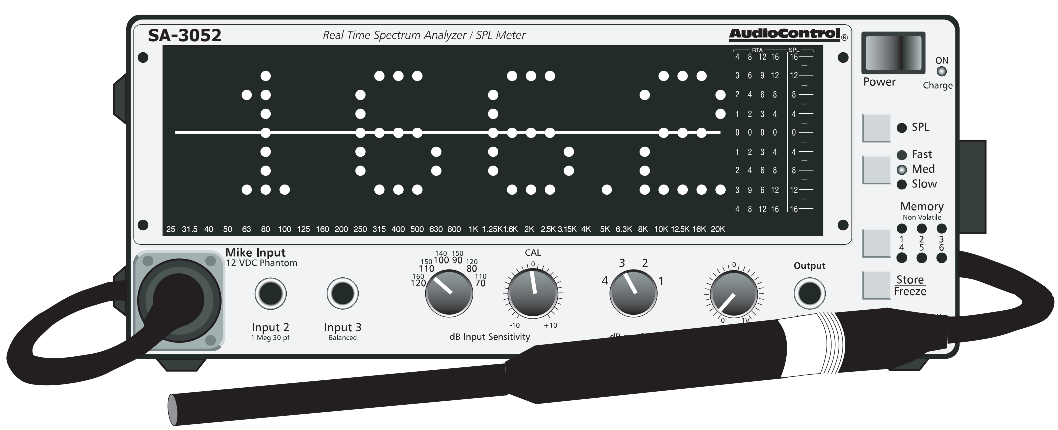
If all else fails, try another printer cable or try using the cable on a PC-type computer. Cables can go bad once in a while.

THERE IS NO SCORING ON THE PRINTOUT

The SA-3052A is not in the competition scoring mode. Turn the analyzer OFF, While turning the SA-3052A back on, press & hold the PRINT switch. Release the PRINT switch when IASCA or USAC is displayed on the analyzer.

OTHER GOODIES FROM AUDIOCONTROL
High SPL option package

This high SPL option is just what the serious competition organizer needs for unlimited SPL events. This high SPL option features a high level microphone, a software upgrade for the SA-3052A and updates to the front-end circuitry. It allows SPL readings over 170dB.



The SA-3052A has many applications. Some of these include:

- Sound System Equalization
- Monitor System Feedback Control
- Movie Theater Setup
- Home Theater Equalization and Setup
- Sound and Music Monitoring
- Component Testing

Sound System Equalization

Sound reinforcement systems, large or small benefit even more from carefully applied equalization. While only dynamite can really cure a really bad case of poor acoustics, equalization comes right after loudspeaker array design when it comes to getting the most out of any speaker system in any room. If you set up in different venues night after night, the SA-3052A can be the impartial judge that helps you make your sound consistent, night after night.

The SA-3052A can work wonders on a home stereo. Most home stereos can benefit from a bit of carefully applied equalization. Such equalization can go a long way towards enhancing your listening pleasure. The SA-3052A can also be used to verify the performance of various portions of the electronics in your system.

Whether you're equalizing a home stereo, or a 100,000 watt concert sound system, the basic technique is the same:

1. Listen
2. Measure
3. Balance speaker components and equalize
4. Listen
5. Trim equalization settings

Hi-Fi Equalization

Before attempting any corrective equalization, use the SA-3052A to make a preliminary assessment of the situation. Use several microphone positions, finally drawing a curve that represents the average of a representative sample of the positions. We recommend no less than 3 different positions in a one to three foot cube around your listening position.

Next, adjust the various level controls on the speakers themselves to smooth out the mid- and high-frequency response. You can combat peaky bass response by trying various elevations above the floor. Conversely, you can aid anemic bass response by using floor or corner placement.

The name of the game here is to get things as "right" as possible without equalization.

Now it's time to use the equalizer. Remember what we discussed in Section 2. It's important to remember to not try to equalize the dips in the overall response curves unless they are quite shallow. Deeper dips may be due to a door, window, or other surface acting as a diaphragmatic absorber (the acoustical equivalent of a black hole). If that is the case, then structural modification of the offending surface is the only real cure. If you get really desperate, there's always explosives.

If you're using an octave-band equalizer, you've probably noticed that the analyzer has a few more bands than the equalizer does. With octave-band equalizers and octave-band analyzers, the important thing that is missing is the response characteristic in-between adjacent bands. The equalizer's controls interact with each other, allowing you to affect the response in the sidebands. If the EQ controls didn't interact, the equalizer wouldn't be very useful...let alone musical. The SA-3052A shows you what is going on in-between the bands of the equalizer and the effects of the controls away from their band-centers. Because of the interaction, you can still deal with problems that aren't conveniently centered on a particular control by splitting the difference between two adjacent controls.

Regardless of which equalizer you're using, an overall guiding principle is: "Less is Better." With good components to start with, all that should be needed is gentle, gradual shaping. Avoid large amounts of boost or cut as well as sharp discontinuities in the overall curve.

Once you're finished, you'll want to add a controlled amount of high-frequency roll-off. Why? Because "ruler-flat" speakers sound unnaturally bright to most ears. Ordinarily we hear most sounds at a distance. This causes high-frequency loss because of the friction between the sound wave and the air, as well as other losses due to absorption caused by walls and other surfaces. As a starting point, try 1 to 3 dB per octave, starting somewhere between 1 and 8 kHz. Experiment; let your ears be your guide and the SA-3052A be your compass.

Sound System Equalization

Equalizing a large sound system isn't much different from equalizing a stereo system. Granted, the scale of things is quite different, but the process is more or less the same.

One thing that you need to keep in mind when equalizing a system in a large space is that the equalized response curve represents the average response taken at many points within the room. This alone can make the equalization process quite tricky. The aiming of the components of the array, their mechanical alignment, and their directional characteristics all contribute. It's easy to make the system curve look right at one point, but how about the other seats in the room?

One source of problems is picking a microphone position that is too close to the loudspeakers. If the loudspeaker has flat power response throughout its coverage pattern, then you are safe. If not, then what happens is that you end up equalizing the direct field response, which is fine at that location but since the power response isn't flat, you get a lumpy response curve at other locations. On the other hand, if you equalize outside of the direct field (in the reverberant field) and successfully flatten things out then the direct field will be lumpy. The solution is constant coverage loudspeakers that maintain their directional characteristics over a wide frequency range.

Monitor Speaker Equalization

Monitors in this context are the monitors used in a recording studio for monitoring and evaluating the program being recorded. Equalization in this context is especially critical because any response aberration in the control room monitor speakers has an inverse relationship on the material being monitored once the material is removed from the room and played on other equipment.

Consider: Your playback system has a midrange bump in it around 3kHz, and a fairly flat high-end. You're mixing some material down using these speakers as a reference. When the producer takes the material out of the studio and listens to it that night at home, he notes that the vocals didn't seem quite as present as they were at the studio and that the drums seem a bit lifeless.

What has happened is that the 3k bump caused you pull the vocals down in the mix (since most vocal energy is around 3k) and the flat high-end resulted in not having enough treble energy content in the final mix. This occurs partially from a level judgment error (too much top end in the monitors, not enough on tape) and perhaps from an equalization error (it sounded fine in the monitors, why EQ it?).

If the speakers have a bass bump, then your end-product could be bass shy. As you can see, the speakers really do have an inverse effect on your end-product.

Careful equalization of your monitor system using the SA-3052A and a good equalizer can give your studio the ability to make tapes that sound good anywhere. The process and pitfalls are very similar to equalizing a home hi-fi system. Additionally you should be aware of the following:

- Your microphone positions should be approximately where the engineer sits, at approximately head level. It's not a bad idea to use several positions at approxi mately ear level, but along the length of the console. Make sure that the response at the producer's chair is very similar to that at the engineer's chair.
- Make the overall response curve smooth rather than flat.
- Add some controlled rolloff at the high frequencies to compensate for the rolled-off high-frequency response of most home systems.
- Pay attention to the response in the midrange of the audio spectrum, between 300 Hz and 5KHz. Make sure that it is smooth and reasonably flat.
- Do flatten peaks.
- Do remove shallow, wide bandwidth holes.
- Do not remove large holes of any bandwidth.
- Beware of adding large amounts of boost at low frequencies. Unless you have infinite amplifier power and speakers capable of moving infinitely far, this is an invitation to amplifier clipping or speaker destruction.
- Remember that 3 dB of boost requires twice the amplifier power at that frequency.

Feedback tuning

What could be simpler? Make the sound system feed back, find the feedback frequency on the equalizer and notch it out slightly. Repeat as necessary. One-third octave equalizers make this a tricky proposition to say the least.

Before the SA-3052A, the process of finding the feedback frequency on the equalizer was the hard part. Some of the techniques used are: guess-and-by-gosh; using a frequency counter; zero-beat with an oscillator; and relating the feedback pitch to the nearest musical pitch, then converting the musical pitch to a numerical frequency (easier than it sounds).

With the SA-3052A connected across the mixer output (you can use the microphone instead), bring the system slowly into feedback. Watch the display. When the system starts feeding back, reduce the gain. The offending frequency is the last one to decay off of the screen. This even works during performance. Now find the frequency on your equalizer and dial in some cut at that frequency. Once more, bring the system slowly into feedback. Repeat the squeal and notch technique until two or more frequencies feed back simultaneously.

Component Checkout

You can use the SA-3052A to check out the frequency response of any component that you can get pink noise into or out of. Of course, signal processors, preamplifiers and amplifiers can all use the pink noise output of the SA-3052A.

Signal sources require a pink noise source compatible with their input format. For instance, a turntable requires a pink noise record, a tape machine requires a pink noise tape. Pink noise sources are available for all common high-fi sources, including CD players.

The diagrams in Figures 5.1 to 5.6 show how to wire everything. The 1dB/step setting gives maximum resolution. Most purely electronic devices should be able to easily exceed the 1dB resolution of the SA-3052A display. Most tape machines (especially cassette machines) will not.

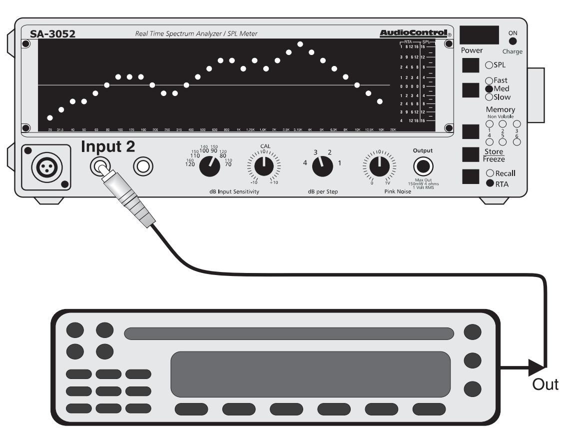


Figure 5.1. Checking a signal source

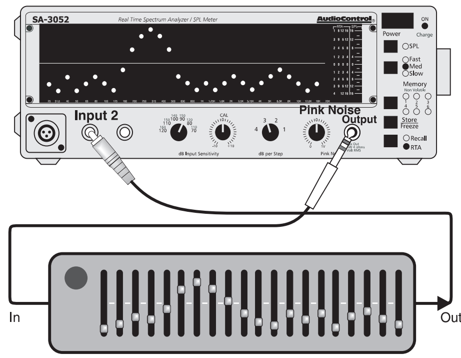


Figure 5.2. Checking a signal processor or preamp

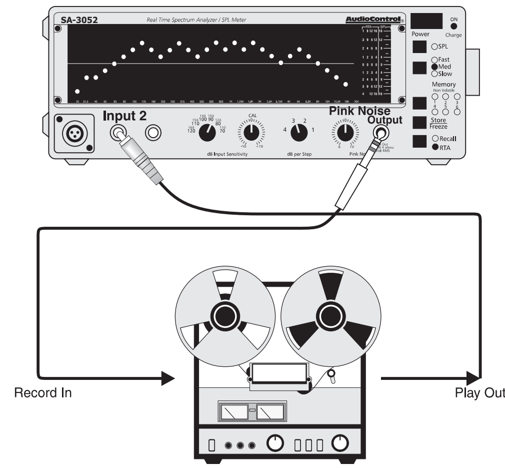


Figure 5.3. Checking a tape recorder

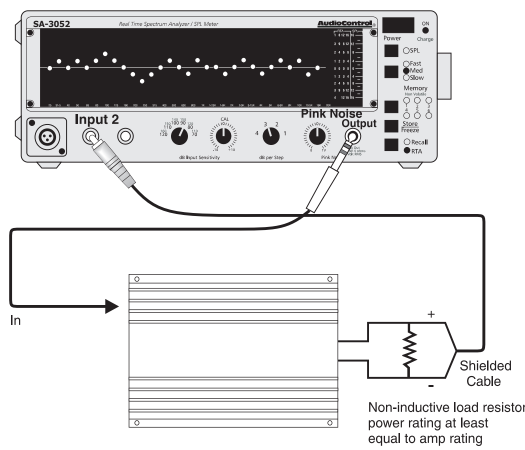


Figure 5.4. Checking a power amplifier

Sound and Music Monitoring

Many recording studios use a real-time analyzer as a music monitor. More than a light show, the analyzer tells the engineer many things about its input signal. Besides instantaneous amplitude, the display shows the spectral balance of the input signal. This is useful in deciding how to equalize a certain instrument, or in trying to figure out how another engineer equalized a certain instrument.

Likewise, if the mix is starting to sound cluttered, a glance at the RTA can tell if the problem is energy build-up in some portion of the frequency spectrum. Conversely, the RTA can identify a portion of the spectrum that is perhaps under-used. The engineer may elect to equalize an instrument to place the dominant portion of its energy content into that region. This helps to make the instrument heard, while not adding clutter to the mix. Of course, the equalization picked should favor the instrument. You might look at the problem as picking the instrument to fit the equalization. It may seem like the chicken before the egg, but it works!

Crossover Testing

Our last application is crossover testing. Actually, this is just an extension of the setups used for component checking. The setup is shown in Figure 5.5. You'll need the following items:

1. Two load resistors, resistance equal to the impedance rating of the crossover. 1 Watt power rating is sufficient. You can also use the speakers that will be connected to the crossover as a load. If you're checking an active crossover, you don't need the load resistors. Use the setup shown in Figure 5.6.

2. The SA-3052A.

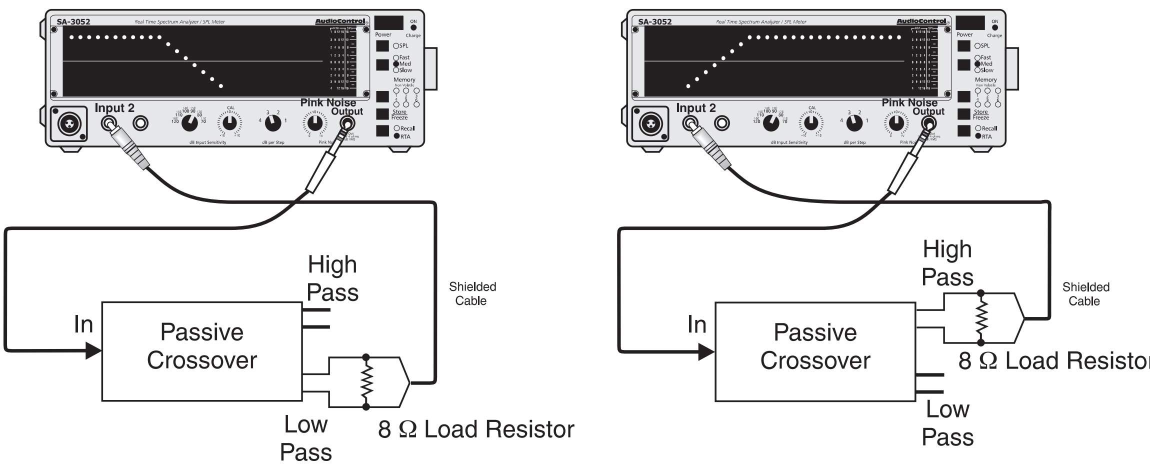


Figure 5.5. Checking a passive crossover

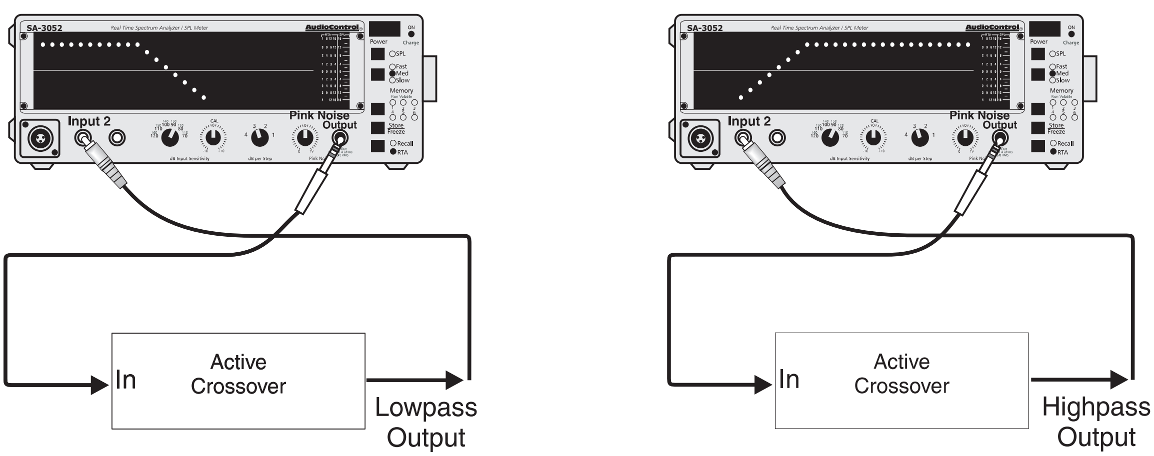


Figure 5.6. Checking an active crossover

It really doesn't matter which output you test first. Set the pink noise output of the SA-3052A at maximum. Set the SA-3052A for a display reading about 3 LED's below the top of the display. Use the 1 dB/step setting. Count three LED's down from the flat portion of the line shown on the display. Move to the right or left until you intersect the falling portion of the line shown on the display. Now move straight down to read the approximate crossover frequency. Store this reading in one of the memories.

Now repeat the operation for the other crossover output. Store this reading also. Now toggle between the two memories. The two curves should look reasonably symmetrical (mirror images of each other). Beware, however, some crossover designs are deliberately asymmetrical.

If you're using load resistors for testing, note that the load that is normally connected to the crossover looks only vaguely like a resistor. It's a good idea to test crossovers both ways: using a dummy load resistor (ideal conditions) and using the actual loudspeaker loads (real-world conditions).

This section of the SA-3052A manual discusses the theory of operation to a functional block diagram level. This section is intended to familiarize you with the major circuit blocks that make up the SA-3052A's circuitry.

The circuitry of the SA-3052A is contained on two printed circuit boards (PCBs). The analog circuitry and part of the digital circuitry are contained on the main PCB, while the microprocessor and it's related circuitry is contained on a second PCB.

The discussion begins at the input connectors to the SA-3052A.

Input Circuitry

Microphone level signals are applied to the preamplifier circuit. The preamplifier consists of three op-amps connected as an instrumentation amplifier. This converts the balanced input signal into an unbalanced signal, and supplies enough gain to bring the input signal up to the SA-3052A's standard operating level. The overall gain of the preamplifier circuit is set by the **dB Input switch**. Simplex (phantom) powering for 12 volt condenser microphones is also provided at the microphone input connector.

Line level signals applied to the ¼" tip-ring-sleeve phone jack are first attenuated to microphone level, and then sent to the preamplifier circuit. Signals applied to the RCA connector are applied to an op-amp that provides any voltage gain needed as well as impedance buffering. The overall gain is set by the **dB Input switch**.

The output signals from both preamplifiers are then summed together. The gain of the summing amplifier is also set by the **dB Input switch**. The last stage of the input circuit is another variable gain stage. The gain of this stage is also controlled by the **±10dB control**. Lastly, the signal is scaled to the correct level by the **dB/Step switch**.

One-Third Octave Filters

After the input circuitry, the signal drives the inputs of thirty one-third octave filters and the broadband SPL circuit. Since the SPL circuit is simpler, we'll discuss it first.

The input to the SPL circuit is a broadband signal. It is rectified, filtered, and applied to one input of the multiplexer (see the next section).

Each of the thirty one-third octave filters is the same, except for part values. Each of the filters is made up of two cascaded, two-pole, bandpass filters. The Q (bandwidth) and center frequency of each filter is precisely set by precision resistors and capacitors. The center frequencies of the two filters are split on each side of the ANSI specified center frequencies. The proper combination of filter Q and the offset center frequencies combines to give an overall curve that meets the ANSI specification for a one-third octave, Class II, type E filter.

Each of the thirty filter outputs is then rectified, filtered, and sent on to the multiplexer.

Signal Multiplexing and Analog to Digital Conversion

The multiplexer switches between each filter output, in sequence, for a fraction of a second per filter. Each filter output is scanned for the same period of time. Another multiplexer input reads the output of the broadband SPL circuit discussed earlier.

The output of the multiplexer contains the sequential output of each of the thirty analyzer filters and the SPL circuit output. This signal is then sent to the analog to digital (A/D) converter.

The A/D converter digitizes the analog output of the multiplexer into a digital word whose value represents the value of its input signal. Since the input of the multiplexer contains 31 different channels, the A/D converter repeats the conversion process 31 times, once for each of the multiplexer inputs.

Microprocessor Circuitry

The microprocessor (µP) controls the timing of various events within the SA-3052A, writes information into its memory, and retrieves that information when needed. In addition, the µP reads the settings of various switches and maintains the status of the various LED indicators on the front panel. The µP program is stored in a 512K EPROM (Erasable Programmable Read Only Memory). A 2K CMOS RAM (Random Access Memory) provides memory for the six display memories, and for scratch-pad memory used during instrument operation.

Periodically, the microprocessor instructs the A/D converter to make another conversion and then place the result on the internal data bus. The µP reads this data, and uses it to drive the display and/or memory. Once the µP has the data from the A/D converter, it performs some math on it (decibel conversion), sends it to the display, and stores it in the temporary display memory.

When told to store a reading, the µP transfers the contents of the temporary display memory to another RAM location for safekeeping. On recall, the µP does the opposite, moving the data from RAM to the temporary display memory.

LED Display

The SA-3052A display gets its data from the µP bus. The display is row and column multiplexed to conserve power. Eight-bit latches hold the row and column data for each column in the display. The intersection of a selected row and selected column results in an illuminated LED at that position of the selected column. A constant-current power supply ensures consistent brightness across the display.

Pink Noise Generator

The pink noise generator uses a series of digital shift registers, connected as a pseudo-random sequence generator. The output of this generator looks like (and for all practical purposes is) white noise.

The output of the white noise generator drives the pinking filter. This filter applies a precise 3 dB/octave rolloff to its input signal. The resultant output is pink noise.

The output of the pink noise generator is a 150 mW audio power amplifier IC. The output amplifier has sufficient output current capability to drive directly into a speaker-impedance (4 ohm) load.

The WARRANTY

People are scared of warranties. Lots of fine print. Months of waiting around. Well, fear no more. This warranty is designed to make you rave about us to your friends. It's a warranty that looks out for you and helps you resist the temptation to have your friend, who's "good with electronics", try to repair your AudioControl product. So go ahead, read this warranty, then take a few days to enjoy your new SA-3052A equalizer before sending in the warranty card and comments.

"Conditional" doesn't mean anything ominous. The Federal Trade Commission tells all manufacturers to use the term to indicate that certain conditions have to be met before they'll honor the warranty. If you meet all of these conditions, we will warrant all materials and workmanship on the SA-3052A equalizer for one (1) year from the date you bought it, and we will fix or replace it, at our option, during that time.

Here are the conditional conditions:

1. You have to fill out the warranty card and send it to us within 15 days after purchasing the SA-3052A equalizer.

2. You must keep your sales receipt for proof of purchase showing when and from whom the unit was bought. We're not the only ones who require this, so it's a good habit to get into with any major purchase.

3. The SA-3052A equalizer must have originally been purchased from an authorized AudioControl dealer. You do not have to be the original owner, but you do need a copy of the original sales slip.

4. You cannot let anybody who isn't: (A) the AudioControl factory; or (B) somebody authorized in writing by AudioControl to service the SA-3052A equalizer. If anyone other than (A), or (B) messes with the SA-3052A equalizer, that voids your warranty.

5. The warranty is also void if the serial number is altered or removed, or if the SA-3052A equalizer has been used improperly. Now that sounds like a big loophole, but here is all we mean by it:

Unwarranted abuse is: (A) physical damage (don't use the SA-3052A equalizer to level your projection TV); (B) improper connections (120 volts into the RCA jacks can fry the poor thing); (C) sadistic things. This is the best product we know how to build, but if you strap it to the front bumper of your Range Rover, something might break.

Assuming you conform to 1 through 5, and it really isn't all that hard to do, we get the option of fixing your old unit or replacing it with a new one.

Legalese Section

This is the only warranty given by AudioControl. This warranty gives you specific legal rights, and you may also have rights that vary from state to state. Promises of how well the SA-3052A equalizer will work are not implied by this warranty. Other than what we've said we'll do in this warranty, we have no obligation, express or implied. We make no warranty of merchantability or fitness for any particular purpose. Also neither we nor anyone else who has been involved in the development or manufacture of the unit will have any liability of any incidental, consequential, special or punitive damages, including but not limited to any lost profits or damage to other parts of your system by hooking up to the unit. Whether the claim is one for breach of warranty, negligence of other tort, or any other kind of claim. Some states do not allow limitations of consequential damages.

Failure to send in a properly completed warranty card negates any service claims.

The warranty included with the unit shall supersede this plain-text version, if there is any inconsistency between the two.

What to do if you need service

First, contact AudioControl, either by phone 425/775-8461 or FAX 425/778-3166. We'll verify if there is anything wrong that you can fix yourself, or arrange to have it sent back to our factory for repair. Please include the following items with the returning unit:

1) A copy of your proof of purchase (that sales receipt we've been harping about). No originals please. We cannot guarantee returning them to you.
2) A brief explanation of the trouble you are having with the SA-3052A. (You'd be surprised how many people forget this.)
3) A return street address. (No PO Boxes, please)
4) A daytime phone number in case our tech has a question about the problem you are having.

You're responsible for the freight charges to us, but we'll pay the return freight back. We match whatever shipping method you send it to us, so if you return the unit overnight freight, we send it back overnight. We recommend United Parcel Service (UPS) for most shipments.

Service Information

WARNING: There are no user serviceable parts inside the SA-3052A. Lethal voltages are present inside the case. Refer all servicing to qualified service personnel.

Repair service is available at:

AudioControl®
Attn: Service Department
22410 70th Avenue West
Mountlake Terrace, WA 98043 USA
(425)775-8461 • FAX (425)778-3166
e-mail: service@audiocontrol.com

Specifications
SA-3052A One-third Octave Real-Time Audio Analyzer

Instrument Type .... ANSI Class II real-time audio analyzer

Inputs

Microphone .... Balanced low impedance
Connector .... 3 pin XLR female
Impedance .... Suitable for 150 ohm microphones, actual impedance is 5 Kohms
Phantom Power .... +12V simplex
Acoustical Signal Level .... 44 dB SPL to 136 dB SPL
Instrumentation .... High-impedance, unbalanced input
Connector .... RCA
Impedance .... 1 Mohm in parallel with 30 pF
Signal Level .... -56 to +36 dBu
Balanced Bridging .... Balanced, high-impedance input
Connector .... ¼” tip-ring-sleeve phone jack
Impedance .... 10 Kohms
Signal Level .... -56 to +36 dBu

Input Calibration

Microphone .... -30 dBu = 94 dB SPL = 24.5 mVrms
Instrumentation .... 0 dBu = 100 dB SPL = 0.775 Vrms
Balanced Bridging .... 0 dBu = 100 dB SPL = 0.775 Vrms

Filters

Filter Characteristic .... Fourth-order bandpass. Meets or exceeds ANSI S1.11-1986 ClassII, Type E specifications
Frequency Range .... 30 1/3 octave bands from 25 to 20 kHz on ISO preferred 1/3 octave center frequencies
Center Frequency Accuracy .... 2% of design center
Band Center Frequencies .... 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500, 16000, 20000 Hz

Display

Spectrum Display .... 9 x 30 LED matrix
SPL Display .... 9 LED bargraph (4dB resolution)
4 digit full-screen display ($1/10$th dB resolution)
Display Resolution .... 1, 2, 3, or 4dB per LED
Display Speed .... Fast, Medium, Slow and 20 second average

Memory

Type .... CMOS SRAM with lithium battery backup
Number .... Six

Physical

Dimensions .... Smaller than a breadbox (4.1”h x 10”w x 12.75”d)
Weight .... 11.5 pounds

Electrical

Power Requirements .... Line Voltage: 108-125VAC, 60 Hz. 208-240VAC, 50/60Hz
Power Consumption .... 15 watts

In the interest of continuing product development, specifications of AudioControl products are subject to change without prior notice.

AudioControl®

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